



Administrator's Guide for the SoundPoint® IP/SoundStation® IP Family

SIP 3.0.2

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4750 Willow Road
Pleasanton, CA 94588-2708
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About This Guide

The Administrator's Guide for the SoundPoint IP / SoundStation IP family is for administrators who need to configure, customize, manage, and troubleshoot SoundPoint IP / SoundStation IP phone systems. This guide covers the SoundPoint IP 301, 320, 330, 430, 501, 550, 560, 600, 601, and 650 desktop phones, and the SoundStation IP 4000 conference phone.

The following related documents for SoundPoint IP / SoundStation IP family are available:

- Quick Start Guides, which describe how to assemble the phones
- Quick User Guides, which describe the most basic features available on the phones
- User Guides, which describe the basic and advanced features available on the phones
- Developer's Guide, which assists in the development of applications that run on the SoundPoint IP / SoundStation IP phone's Microbrowser
- Technical Bulletins, which describe workarounds to existing issues
- Release Notes, which describe the new and changed features and fixed problems in the latest version of the software

For support or service, please contact your Polycom® reseller or go to Polycom Technical Support at <http://www.polycom.com/support/voice/>.

Polycom recommends that you record the phone model numbers, software (both the bootROM and SIP), and partner platform for future reference.

SoundPoint IP / SoundStation IP models: _____

BootROM version: _____

SIP Application version: _____

Partner Platform: _____

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Introducing the SoundPoint IP / SoundStation IP Family

This chapter introduces the SoundPoint IP / SoundStation IP family, which is supported by the software described in this guide.

The SoundPoint IP / SoundStation IP family provides a powerful, yet flexible IP communications solution for Ethernet TCP/IP networks, delivering excellent voice quality. The high-resolution graphic display supplies content for call information, multiple languages, directory access, and system status. The SoundPoint IP / SoundStation IP family supports advanced functionality, including multiple call and flexible line appearances, HTTPS secure provisioning, presence, custom ring tones, and local conferencing.

The SoundPoint IP / SoundStation IP phones are end points in the overall network topology designed to interoperate with other compatible equipment including application servers, media servers, internet-working gateways, voice bridges, and other end points

The following models are described:

- [SoundPoint IP Desktop Phones](#)
 - IP 301
 - IP 320/330
 - IP 430
 - IP 501
 - IP 550/560
 - IP 600/601
 - IP 650
- [SoundStation IP Conference Phone](#)
 - IP 4000

This chapter also lists the key features available on the SoundPoint IP / SoundStation IP phones running the latest software.

SoundPoint IP Desktop Phones

This section describes the current SoundPoint IP desktop phones. For individual guides, refer to the product literature available at <http://www.polycom.com/support/voice/>. Additional options are also available. For more information, contact your Polycom distributor.

The currently supported desktop phones are:

- SoundPoint IP 301



- SoundPoint IP 320/330



- SoundPoint IP 430



- SoundPoint IP 501



- SoundPoint IP 550/560



- SoundPoint IP 600/601



- SoundPoint IP 650



SoundStation IP Conference Phone

This section describes the current SoundPoint IP conference phone. For individual guides, refer to the product literature available at <http://www.polycom.com/support/voice/>. Additional options are also available. For more information, contact your Polycom distributor.

The currently supported conference phone is :

- SoundStation IP 4000



Key Features of Your SoundPoint IP / SoundStation IP Phones

The key features of the SoundPoint IP / SoundStation IP phones are:

- Award winning sound quality and full-duplex speakerphone or conference phone
 - Permits natural, high-quality, two-way conversations (one-way, monitor speaker in the SoundPoint IP 301)
 - Uses Polycom's industry leading Acoustic Clarity Technology
- Easy-to-use
 - An easy transition from traditional PBX systems into the world of IP
 - Up to 18 dedicated hard keys for access to commonly used features
 - Up to four context-sensitive soft keys for further menu-driven activities
- Platform independent
 - Supports multiple protocols and platforms enabling standardization on one phone for multiple locations, systems and vendors
 - Polycom's support of the leading protocols and industry partners makes it a future-proof choice
- Field upgradeable
 - Upgrade SoundPoint IP / SoundStation IP as standards develop and protocols evolve
 - Extends the life of the phone to protect your investment

- Application flexibility for call management and new telephony applications
- Large LCD
 - Easy-to-use, easily readable and intuitive interface
 - Support of rich application content, including multiple call appearances, presence and instant messaging, and XML services
 - 4 line x 20 character monochrome LCD for the SoundPoint IP 301
 - 102 x 23 pixel graphical LCD for the SoundPoint IP 320/330
 - 160 x 80 pixel graphical grayscale LCD for the SoundPoint IP 501
 - 320 x 160 pixel graphical grayscale LCD for the SoundPoint IP 550/560/600/601/650 (supports Asian characters)
 - 248 x 68 pixel graphical LCD for the SoundStation IP 4000
- Dual auto-sensing 10/100baseT Ethernet ports
 - Leverages existing infrastructure investment
 - No re-wiring with existing CAT 5 cabling
 - Simplifies installation
- Power over Ethernet (PoE) port
 - Unused pairs on Ethernet port pairs are used to deliver power to the phone via a wall adapter allowing fewer wires to desktop
 - Optional accessory cable for CiscoR Inline Powering and IEEE 802.3af on the SoundPoint IP 301 and SoundPoint IP 501
 - Built-in PoE on the SoundPoint IP 550, 560, 600, 601, and 650 (auto-sensing)
- Multiple language support
 - Set on-screen language to your preference. Select from Chinese, Danish, Dutch, English, French, German, Italian, Japanese, Korean, Norwegian, Portuguese, Russian, Spanish, and Swedish

Note

In SIP 3.0, default support for Chinese, Japanese, and Korean was removed from the SoundPoint IP 600 and 601.

Overview

This chapter provides an overview of the Session Initiation Protocol (SIP) application and how the phones fit into the network configuration.

SIP is the Internet Engineering Task Force (IETF) standard for multimedia conferencing over IP. It is an ASCII-based, application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints. Like other voice over IP (VoIP) protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

For the SoundPoint IP / SoundStation IP phones to successfully operate as a SIP endpoint in your network, it must meet the following requirements:

- A working IP network is established.
- Routers are configured for VoIP.
- VoIP gateways are configured for SIP.
- The latest (or compatible) SoundPoint IP / SoundStation IP phone SIP application image is available.
- A call server is active and configured to receive and send SIP messages.

For more information on IP PBX and softswitch vendors, go to <http://www.polycom.com/techpartners1/>.

This chapter contains information on:

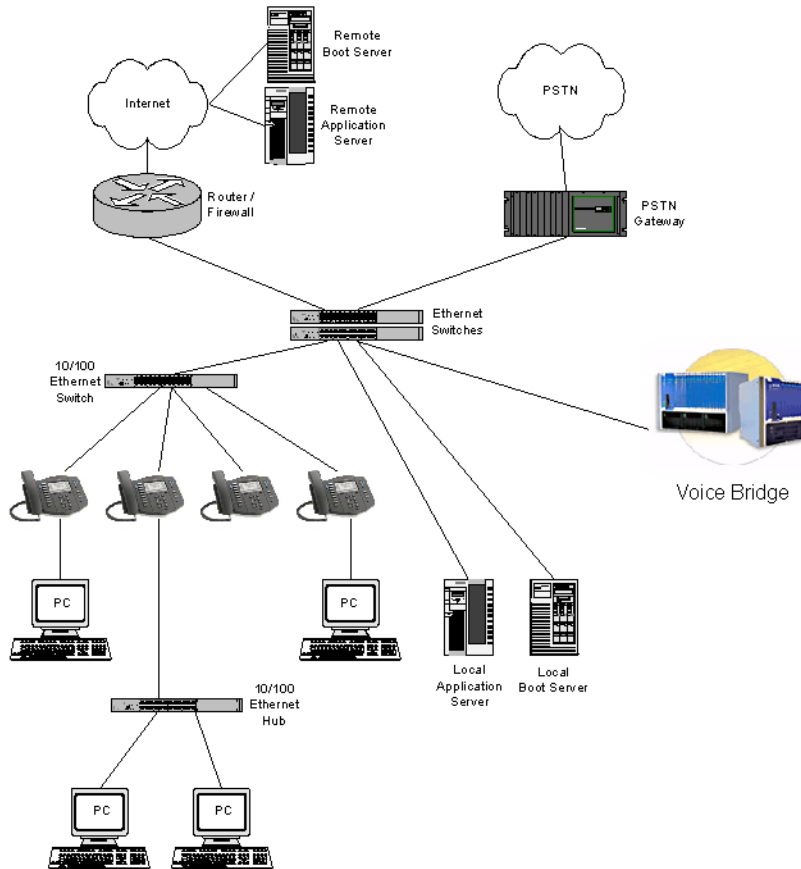
- [Where SoundPoint IP / SoundStation IP Phones Fit](#)
- [Session Initiation Protocol Application Architecture](#)
- [Available Features](#)

To install your SoundPoint IP / SoundStation IP phones on the network, refer to [Setting up Your System](#) on page 3-1. To configure your SoundPoint IP / SoundStation IP phones with the desired features, refer to [Configuring Your](#)

[System](#) on page 4-1. To troubleshoot any problems with your SoundPoint IP / SoundStation IP phones on the network, refer to [Troubleshooting Your SoundPoint IP / SoundStation IP Phones](#) on page 5-1.

Where SoundPoint IP / SoundStation IP Phones Fit

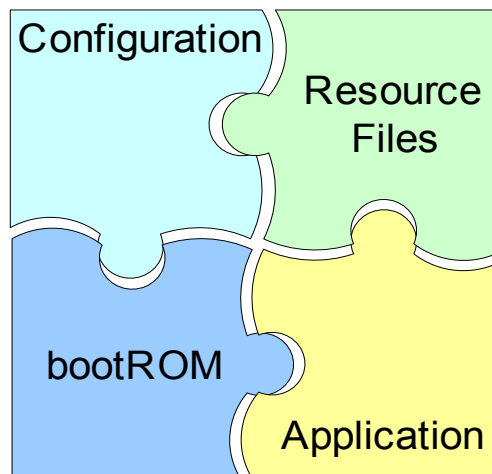
The phones connect physically to a standard office twisted-pair (IEEE 802.3) 10/100 megabytes per second Ethernet LAN and send and receive all data using the same packet-based technology. Since the phone is a data terminal, digitized audio being just another type of data from its perspective, the phone is capable of vastly more than traditional business phones. As SoundPoint IP / SoundStation IP phones run the same protocols as your office personal computer, many innovative applications can be developed without resorting to specialized technology.



Session Initiation Protocol Application Architecture

The software architecture of SIP application is made of 4 basic components:

- **BootROM** – loads first when the phone is powered on
- **Application** – software that makes the device a phone
- **Configuration** – configuration parameters stored in separate files
- **Resource Files** – optional, needed by some of the advanced features



BootROM

The bootROM is a small application that resides in the flash memory on the phone. All phones come from the factory with a bootROM pre-loaded.

The bootROM performs the following tasks in order:

1. Performs a power on self test (POST).
2. (Optional) Allows you to enter the setup menu where various network provisioning options can be set.

The bootROM software controls the user interface when the setup menu is accessed.

3. Requests IP settings and accesses the boot server to look for any updates to the bootROM application.

If updates are found, they are downloaded and saved to flash memory, eventually overwriting itself after verifying the integrity of the download.

4. If a new bootROM is downloaded, format the file system clearing out any application software or configuration files that may have been present.

5. Download the master configuration file.

This file is either called **<MAC-address>.cfg** or **000000000000.cfg**. This file is used by both the bootROM and the application for a list of other files that are needed for the operation of the phone.

6. Examine the master configuration file for the name of the application file, and then look for this file on the boot server.

If the copy on the boot server is different than the one stored in flash memory or, if there is no file stored in flash memory, the application file is downloaded.

Note

If the Application is any SIP version prior to 1.5, the bootROM will also download all the configuration files that are listed in the master configuration file.

7. Extract the application from flash memory.
8. Install the application into RAM, then upload a log file with events from the boot cycle.

The bootROM will then terminate, and the application takes over.

Application

The application manages the VoIP stack, the digital signal processor (DSP), the user interface, and the network interaction. The application managed everything to do with the phone's operation.

The application is a single file binary image and, as of SIP 1.5, contains a digital signature to prevent tampering or loading of rogue software images.

Warning

If your phones are using bootROM 3.0 or later, the application must be signed.

All SIP 1.5 applications and later are signed, but later patched versions of 1.3 and 1.4 support this feature. Refer to the latest *Release Notes* to verify if the image is signed.

There is a new image file in each release of software.

The application performs the following tasks in order:

1. Downloads system and per-phone configuration files and resource files.

These files are called **sip.cfg** and **phone1.cfg** by default. You can customized the filenames.

Note

If the Application is any SIP version prior to 1.5, the bootROM would have downloaded all the configuration files that are listed in the master configuration file.

2. Controls all aspects of the phone after it has restarted.
3. Uploads log files.

BootROM and Application Wrapper

Both the bootROM and the application run on multiple platforms (meaning all previously released versions of hardware that are still supported).

The file stored on the boot server is a wrapper, with multiple hardware specific images contained within. When a new bootROM or application is being saved, the file is read until a header matching the hardware model and revision are found, and then only this image is saved to flash memory.

Configuration

The SoundPoint IP / SoundStation IP phones can be configured automatically through files stored on a central boot server, manually through the phone's local UI or web interface, or a combination of the automatic and manual methods.

The recommended method for configuring phones is automatically through a central boot server, but if one is not available, the manual method will allow changes to most of the key settings.

The phone configuration files consist of:

- [Master Configuration Files](#)
- [Application Configuration Files](#)

Warning

Configuration files should only be modified by a knowledgeable system administrator. Applying incorrect parameters may render the phone unusable. The configuration files which accompany a specific release of the SIP software must be used together with that software. Failure to do this may render the phone unusable.

Master Configuration Files

The master configuration files can be one of:

- Specified master configuration file
- Per-phone master configuration file
- Default master configuration file

For more information, refer to [Master Configuration Files](#) on page A-2.

Application Configuration Files

Typically, the files are arranged in the following manner although parameters may be moved around within the files and the filenames themselves can be changed as needed. These files dictate the behavior of the phone once it is running the executable specified in the master configuration file.

The application files are:

- Application – It contains parameters that affect the basic operation of the phone such as voice codecs, gains, and tones and the IP address of an application server. All phones in an installation usually share this category of files. Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first. By default, **sip.cfg** is included.
- Per-phone – It contains parameters unique to a particular phone user. Typical parameters include:
 - display name
 - unique addresses

Each phone in an installation usually has its own customized version of user files derived from Polycom templates. By default, **phone1.cfg** is included.

Central Provisioning

The phones can be centrally provisioned from a boot server through a system of global and per-phone configuration files. The boot server also facilitates automated application upgrades, logging, and a measure of fault tolerance. Multiple redundant boot servers can be configured to improve reliability.

In the central provisioning method, there are two major classifications of configuration files:

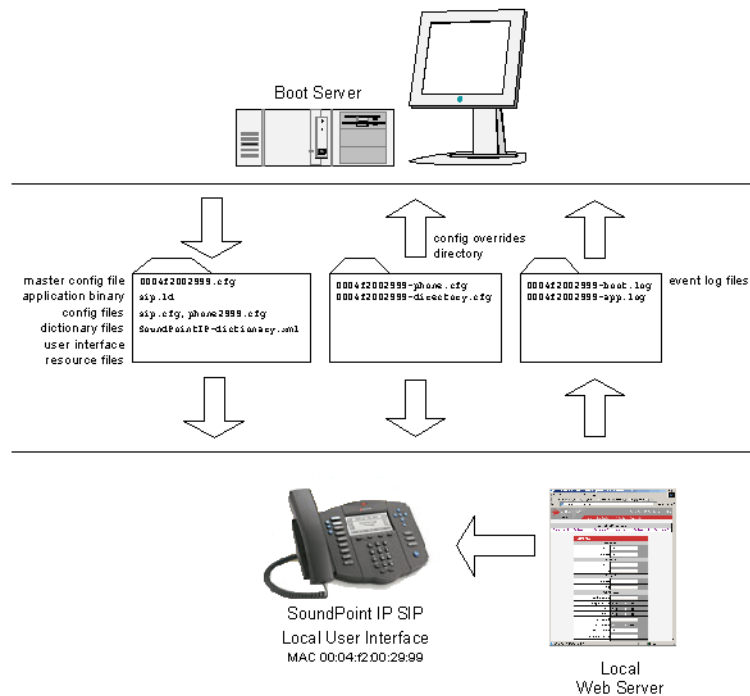
- System configuration files
- Per-phone configuration files

Parameters can be stored in the files in any order and can be placed in any number of files. The default is to have 2 files, one for per-phone setting and one for system settings. The per-phone file is typically loaded first, and could contain system level parameters, letting you override that parameter for a given user. For example, it might be desirable to set the default CODEC for a remote user differently than for all the users who reside in the head office. By adding the CODEC settings to a particular user's per-phone file, the values in the system file are ignored.

Note

Verify the order of the configuration files. Parameters in the configuration file loaded first will overwrite those in later configuration files.

The following figure shows one possible layout of the central provisioning method.



Manual Configuration

When the manual configuration method is employed, any changes made are stored in a configuration override file. This file is stored on the phone, but a copy will also be uploaded to the central boot server if one is being used. When the phone boots, this file is loaded by the application after any centrally provisioned files have been read, and its settings will override those in the centrally provisioned files.

This can create a lot of confusion about where parameters are being set, and so it is best to avoid using the manual method unless you have good reason to do so.

Resource Files

In addition to the application and the configuration files, the phones may require resource files that are used by some of the advanced features. These files are optional, but if the particular feature is being employed, these files are required.

Some examples of resource files include:

- Language dictionaries
- Custom fonts

- Ring tones
- Synthesized tones
- Contact directories

Available Features

Note

The SoundPoint IP 300 and 500 phones will be supported on the latest maintenance patch release of the SIP 2.1 software stream—currently SIP 2.1.2. Any new features introduced after SIP 2.1.2 are not supported.

This section provides information the features available on the SoundPoint IP / SoundStation IP phones:

- Basic Features
 - [Automatic Off-Hook Call Placement](#)—Supports an optional automatic off-hook call placement feature for each .
 - [Call Forward](#)—Provides a flexible call forwarding feature to forward calls to another destination.
 - [Call Hold](#)—Pauses activity on one call so that the user may use the phone for another task, such as making or receiving another call.
 - [Call Log](#)—Contains call information such as remote party identification, time and date, and call duration in three separate lists, missed calls, received calls, and placed calls on most platforms.
 - [Call Park/Retrieve](#)—An active call can be parked. A parked call can be retrieved by any phone.
 - [Call Timer](#)—A separate call timer, in hours, minutes, and seconds, is maintained for each distinct call in progress.
 - [Call Transfer](#)—Call transfer allows the user to transfer a call in progress to some other destination.
 - [Call Waiting](#)—When an incoming call arrives while the user is active on another call, the incoming call is presented to the user visually on the display and a configurable sound effect will be mixed with the active call audio.
 - [Called Party Identification](#)—The phone displays and logs the identity of the party specified for outgoing calls.
 - [Calling Party Identification](#)—The phone displays the caller identity, derived from the network signalling, when an incoming call is presented, if information is provided by the call server.

- **Connected Party Identification** – The identity of the party to which the user has connected is displayed and logged, if the name is provided by the call server.
- **Context Sensitive Volume Control** – The volume of user interface sound effects, such as the ringer, and the receive volume of call audio is adjustable.
- **Customizable Audio Sound Effects** – Audio sound effects used for incoming call alerting and other indications are customizable.
- **Directed Call Pick-Up** and **Group Call Pick-Up** – Calls to another phone can be picked up by dialing the extension of the other phone. Calls to another phone within a pre-defined group can be picked up without dialing the extension of the other phone.
- **Distinctive Call Waiting** – Calls can be mapped to distinct call waiting types.
- **Distinctive Incoming Call Treatment** – The phone can automatically apply distinctive treatment to calls containing specific attributes.
- **Distinctive Ringing** – The user can select the ring type for each line and the ring type for specific callers can be assigned in the contact directory.
- **Do Not Disturb** – A do-not-disturb feature is available to temporarily stop all incoming call alerting.
- **Graphic Display Backgrounds** – A picture or design displayed on the background of the graphic display.
- **Handset, Headset, and Speakerphone** – SoundPoint IP phones come standard with a handset and a dedicated headset connection (not supplied). The SoundPoint IP 320, 330, 430, 500, 501, 550, 560, 600, 601, and 650 phones and SoundStation IP 4000 phone are full-duplex speakerphones. The SoundPoint IP 301 phone is a listen-only speakerphone.
- **Idle Display Animation** – All phones except the SoundPoint IP 301 can display a customized animation on the idle display in addition to the time and date.
- **Last Call Return** – The phone allows call server-based last call return.
- **Local / Centralized Conferencing** – The phone can conference together the local user with the remote parties of two independent calls and can support centralized conferences for which external resources are used such as a conference bridge.
- **Local Contact Directory** – The phone maintains a local contact directory that can be downloaded from the boot server and edited locally.
- **Local Digit Map** – The phone has a local digit map to automate the setup phase of number-only calls.

- [Message Waiting Indication](#) – The phone will flash a message-waiting indicator (MWI) LED when instant messages and voice messages are waiting.
- [Microphone Mute](#) – When the microphone mute feature is activated, visual feedback is provided.
- [Missed Call Notification](#) – The phone can display the number of calls missed since the user last looked at the Missed Calls list.
- [Soft Key Activated User Interface](#) – The user interface makes extensive use of intuitive, context-sensitive soft key menus.
- [Speed Dial](#) – The speed dial system allows calls to be placed quickly from dedicated keys as well as from a speed dial menu.
- [Time and Date Display](#) – Time and date can be displayed in certain operating modes such as when the phone is idle and during a call.
- **Advanced Features**
 - [Automatic Call Distribution](#) – Supports ACD agent available and unavailable and allows ACD login and logout. Requires call server support.
 - [Bridged Line Appearance](#) – Calls and lines on multiple phones can be logically related to each other. Requires call server support.
 - [Busy Lamp Field](#) – Allows monitoring the hook status and remote party information of users through the busy lamp field (BLF) LEDs and displays on an attendant console phone. Requires call server support.
 - [Configurable Feature Keys](#) – Certain key functions can be changed from the factory defaults.
 - [Corporate Directory](#) – The phone can be configured to access your corporate directory if it has a standard LDAP interface.
 - [Customizable Fonts and Indicators](#) – The phone's user interface can be customized by changing the fonts and graphic icons used on the display and the LED indicator patterns.
 - [Downloadable Fonts](#) – New fonts can be loaded onto the phone.
 - [Instant Messaging](#) – Supports sending and receiving instant text messages.
 - [Microbrowser](#) – The SoundPoint IP 430, 501, 550, 560, 600, 601, and 650 phones and the SoundStation IP 4000 phone support an XHTML microbrowser.
 - [Microsoft Live Communications Server 2005 Integration](#) – SoundPoint IP and SoundStation IP phones can be used with Microsoft Live Communications Server 2005 and Microsoft Office Communicator to help improve business efficiency and increase productivity and to share ideas and information immediately with business contacts. Requires call server support.

- [Multilingual User Interface](#) – All phones except SoundPoint IP 301 have multilingual user interfaces.
- [Multiple Call Appearances](#) – The phone supports multiple concurrent calls. The hold feature can be used to pause activity on one call and switch to another call.
- [Multiple Line Keys per Registration](#) – More than one line key can be allocated to a single .
- [Multiple Registrations](#) – SoundPoint IP phones support multiple registrations per phone. (SoundStation IP 4000 supports a single registration.)
- [Network Address Translation](#) – The phones can work with certain types of network address translation (NAT).
- [Presence](#) – Allows the phone to monitor the status of other users/devices and allows other users to monitor it. Requires call server support.
- [Real-Time Transport Protocol Ports](#) – The phone treats all real-time transport protocol (RTP) streams as bi-directional from a control perspective and expects that both RTP end points will negotiate the respective destination IP addresses and ports.
- [Recording and Playback of Audio Calls](#) – Recording and playback allows the user to record any active conversation using the phone on a USB device. The files are date and time stamped for easy archiving and can be played back on the phone or on any computer with a media playback program what supports the **.wav** format.
- [Server Redundancy](#) – Server redundancy is often required in VoIP deployments to ensure continuity of phone service for events where the call server needs to be taken offline for maintenance, the server fails, or the connection from the phone to the server fails.
- [Shared Call Appearances](#) – Calls and lines on multiple phones can be logically related to each other. Requires call server support.
- [Synthesized Call Progress Tones](#) – In order to emulate the familiar and efficient audible call progress feedback generated by the PSTN and traditional PBX equipment, call progress tones are synthesized during the life cycle of a call. Customizable for certain regions, for example, Europe has different tones from North America.
- [Voice Mail Integration](#) – Compatible with voice mail servers.
- Audio Features
 - [Acoustic Echo Cancellation](#) – Employs advanced acoustic echo cancellation for hands-free operation.
 - [Audio Codecs](#) – Supports the standard audio codecs.
 - [Automatic Gain Control](#) – Designed for hands-free operation, boosts the transmit gain of the local user in certain circumstances.

- **Background Noise Suppression** – Designed primarily for hands-free operation, reduces background noise to enhance communication in noisy environments.
- **Comfort Noise Fill** – Designed to help provide a consistent noise level to the remote user of a hands-free call.
- **DTMF Event RTP Payload** – Conforms to RFC 2833, which describes a standard RTP-compatible technique for conveying DTMF dialing and other telephony events over an RTP media stream.
- **DTMF Tone Generation** – Generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad.
- **IEEE 802.1p/Q** – The phone will tag all Ethernet packets it transmits with an 802.1Q VLAN header.
- **IP Type-of-Service** – Allows for the setting of TOS settings.
- **Jitter Buffer and Packet Error Concealment** – Employs a high-performance jitter buffer and packet error concealment system designed to mitigate packet inter-arrival jitter and out-of-order or lost (lost or excessively delayed by the network) packets.
- **Low-Delay Audio Packet Transmission** – Designed to minimize latency for audio packet transmission.
- **Voice Activity Detection** – Conserves network bandwidth by detecting periods of relative “silence” in the transmit data path and replacing that silence efficiently with special packets that indicate silence is occurring.
- **Voice Quality Monitoring** – Generates various quality metrics including MOS and R-factor for listening and conversational quality.
- **Security Features**
 - **Local User and Administrator Privilege Levels** – Several local settings menus are protected with two privilege levels, user and administrator, each with its own password.
 - **Configuration File Encryption** – Confidential information stored in configuration files must be protected (encrypted). The phone can recognize encrypted files, which it downloads from the boot server and it can encrypt files before uploading them to the boot server.
 - **Custom Certificates** – When trying to establish a connection to a boot server for application provisioning, the phone trusts certificates issued by widely recognized certificate authorities (CAs).
 - **Incoming Signaling Validation** – Levels of security are provided for validating incoming network signaling.

For more information on each feature and its associated configuration parameters, see the appropriate section in [Configuring Your System](#) on page 4-1.

Setting up Your System

Your SoundPoint IP / SoundStation IP SIP phone is designed to be used like a regular phone on a public switched telephone network (PSTN).

This chapter provides basic instructions for setting up your SoundPoint IP / SoundStation IP phones. This chapter contains information on:

- [Setting Up the Network](#)
- [Setting Up the Boot Server](#)
- [Deploying Phones From the Boot Server](#)
- [Upgrading SIP Application](#)

Because of the large number of optional installations and configurations that are available, this chapter focuses on one particular way that the SIP application and the required external systems might initially be installed and configured in your network.

For more information on configuring your system, refer to [Configuring Your System](#) on page 4-1. For more information on the configuration files required for setting up your system, refer to [Configuration Files](#) on page A-1.

Note

For installation and maintenance of Polycom SoundPoint IP phones, the use of a boot server is strongly recommended. This allows for flexibility in installing, upgrading, maintaining, and configuring the phone. Configuration, log, and directory files are normally located on this server. Allowing the phone write access to the server is encouraged.

The phone is designed such that, if it cannot locate a boot server when it boots up, it will operate with internally saved parameters. This is useful for occasions when the boot server is not available, but is not intended to be used for long-term operation of the phones.

Setting Up the Network

Regardless of whether or not you will be installing a centrally provisioned system, you must perform basic TCP/IP network setup, such as IP address and subnet mask configuration, to get your organization's phones up and running.

The bootROM application uses the network to query the boot server for upgrades, which is an optional process that will happen automatically when properly deployed. For more information on the basic network settings, refer to [DHCP or Manual TCP/IP Setup](#) on page 3-2.

The bootROM on the phone performs the provisioning functions of downloading the bootROM, the **<Ethernet address>.cfg** file, and the SIP application, and uploading log files. For more information, refer to [Supported Provisioning Protocols](#) on page 3-4.

Basic network settings can be changed during bootROM download using the bootROM's setup menu. A similar menu system is present in the application for changing the same network parameters. For more information, refer to [Modifying the Network Configuration](#) on page 3-5.

DHCP or Manual TCP/IP Setup

Basic network settings can be derived from DHCP, or entered manually using the phone's LCD-based user interface, or downloaded from configuration files.



Polycom recommends using DHCP where possible to eliminate repetitive manual data entry.

The following table shows the manually entered networking parameters that may be overridden by parameters obtained from a DHCP server, an alternate DHCP server, or configuration file:

Parameter	DHCP Option	DHCP	Alternate DHCP	Configuration File (application only)	Local FLASH
		⇔ priority when more than one source exists ⇔			
		1	2	3	4
IP address	1	•	-	-	•
subnet mask	1	•	-	-	•
IP gateway	3	•	-	-	•

Parameter	DHCP Option	DHCP	Alternate DHCP	Configuration File (application only)	Local FLASH
boot server address	Refer to DHCP Menu on page 3-7	•	•	-	•
SIP server address	151 <i>Note: This value is configurable.</i>	•	-	-	•
SNTP server address	42 then 4	•	-	•	•
SNTP GMT offset	2	•	-	•	•
DNS server IP address	6	•	-	-	•
alternate DNS server IP address	6	•	-	-	•
DNS domain	15	•	-	-	•
VLAN ID	Refer to DHCP Menu on page 3-7	Warning: Cisco Discovery Protocol (CDP) overrides Local FLASH that overrides DHCP VLAN Discovery.			

For more information on DHCP options, go to <http://www.ietf.org/rfc/rfc2131.txt?number=2131> or <http://www.ietf.org/rfc/rfc2132.txt?number=2132>.

Note

The configuration file value for **SNTP server address** and **SNTP GMT offset** can be configured to override the DHCP value. Refer to `tcpIpApp.snntp.address.overrideDHCP` in [Time Synchronization <snntp/>](#) on page A-53.

The CDP value can be obtained from a connected Ethernet switch if the switch supports CDP.

In the case where you do not have control of your DHCP server or do not have the ability to set the DHCP options, an alternate method of automatically discovering the provisioning server address is required. Connecting to a secondary DHCP server that responds to DHCP INFORM queries with a requested boot server value is one possibility. For more information, refer to <http://www.ietf.org/rfc/rfc3361.txt?number=3361> and <http://www.ietf.org/rfc/rfc3925.txt?number=3925>.

Supported Provisioning Protocols

The bootROM performs the provisioning functions of downloading configuration files, uploading and downloading the configuration override file and user directory, and downloading the dictionary and uploading log files.

The protocol that will be used to transfer files from the boot server depends on several factors including the phone model and whether the bootROM or SIP application stage of provisioning is in progress. By default, the phones are shipped with FTP enabled as the provisioning protocol. If an unsupported protocol is specified, this may result in a defined behavior (see the table below for details of which protocol the phone will use). The Specified Protocol listed in the table can be selected in the *Server Type* field or the *Server Address* can include a transfer protocol, for example `http://usr:pwd@server` (refer to [Server Menu](#) on page 3-9). The boot server address can be an IP address, domain string name, or URL. The boot server address can also be obtained through DHCP. Configuration file names in the `<Ethernet address>.cfg` file can include a transfer protocol, for example `https://usr:pwd@server/dir/file.cfg`. If a user name and password are specified as part of the server address or file name, they will be used only if the server supports them.

Note

A URL should contain forward slashes instead of back slashes and should not contain spaces. Escape characters are not supported. If a user name and password are not specified, the Server User and Server Password will be used (refer to [Server Menu](#) on page 3-9).

	Protocol used by bootROM	Protocol used by SIP Application
Specified Protocol	301, 320, 330, 430, 501, 550, 560, 600, 601, 650, 4000	301, 320, 330, 430, 501, 550, 560, 600, 601, 650, 4000
FTP	FTP	FTP
TFTP	TFTP	TFTP
HTTP	HTTP	HTTP
HTTPS	HTTP	HTTPS

Note

There are two types of FTP methods—active and passive. As of SIP 1.5 (and bootROM 3.0), the SIP application is no longer compatible with active FTP. At that time, secure provisioning was implemented.

Note

Setting Option 66 to `ftp://192.168.9.10` has the effect of forcing a TFTP download. Using a TFTP URL (for example, `ftp://provserver.polycom.com`) has the same effect.

For downloading the bootROM and application images to the phone, the secure HTTPS protocol is not available. To guarantee software integrity, the bootROM will only download cryptographically signed bootROM or application images. For HTTPS, widely recognized certificate authorities are trusted by the phone and custom certificates can be added (refer to [Trusted Certificate Authority List](#) on page C-1).

Modifying the Network Configuration

You can access the network configuration menu:

- **During bootROM Phase.** The network configuration menu is accessible during the auto-boot countdown of the bootROM phase of operation. Press the **Setup** soft key to launch the main menu.
- **During Application Phase.** The network configuration menu is accessible from the phone's main menu. Select *Menu>Settings>Advanced>Admin Settings>Network Configuration*. Advanced Settings are locked by default. Enter the administrator password to unlock. The factory default password is 456.

Phone network configuration parameters may be modified by means of:

- [Main Menu](#)
- [DHCP Menu](#)
- [Server Menu](#)
- [Ethernet Menu](#)
- [Syslog Menu](#)

Use the soft keys, the arrow keys, the **Select** and **Delete** keys to make changes.

Certain parameters are read-only due to the value of other parameters. For example, if the DHCP Client parameter is enabled, the *Phone IP Addr* and *Subnet Mask* parameters are dimmed or not visible since these are guaranteed to be supplied by the DHCP server (mandatory DHCP parameters) and the statically assigned IP address and subnet mask will never be used in this configuration.

Resetting to Factory Defaults

The basic network configuration referred to in the following sections can be reset to factory defaults using a multiple key combination described in [Multiple Key Combinations](#) on page C-9.

Main Menu

The following configuration parameters can be modified on the main setup menu:

Name	Possible Values	Description
DHCP Client	Enabled, Disabled	If enabled, DHCP will be used to obtain the parameters discussed in DHCP or Manual TCP/IP Setup on page 3-2.
DHCP Menu		Refer to DHCP Menu on page 3-7. Note: Disabled when DHCP client is disabled.
Phone IP Address	dotted-decimal IP address	Phone's IP address. Note: Disabled when DHCP client is enabled.
Subnet Mask	dotted-decimal subnet mask	Phone's subnet mask. Note: Disabled when DHCP client is enabled.
IP Gateway	dotted-decimal IP address	Phone's default router.
Server Menu		Refer to Server Menu on page 3-9.
SNTP Address	dotted-decimal IP address OR domain name string	Simple Network Time Protocol (SNTP) server from which the phone will obtain the current time.
GMT Offset	-13 through +12	Offset of the local time zone from Greenwich Mean Time (GMT) in half hour increments.
DNS Server	dotted-decimal IP address	Primary server to which the phone directs Domain Name System (DNS) queries.
DNS Alternate Server	dotted-decimal IP address	Secondary server to which the phone directs Domain Name System queries.
DNS Domain	domain name string	Phone's DNS domain.
Ethernet		Refer to Ethernet Menu on page 3-11.
EM Power	Enabled, Disabled	This parameter is relevant if the phone gets Power over Ethernet (PoE). If enabled, the phone will set power requirements in CDP to 12W so that up to three Expansion Modules (EM) can be powered. If disabled, the phone will set power requirements in CDP to 5W which means no Expansion Modules can be powered (it will not work).
Syslog		Refer to Syslog Menu on page 3-12.

Note

A parameter value of “???” indicates that the parameter has not yet been set and saved in the phone’s configuration. Any such parameter should have its value set before continuing.

The **EM Power** parameter is only available on SoundPoint IP 601 and 650 phones.

Note

To switch the text entry mode on the SoundPoint IP 330/320, press the #. You may want to use URL or IP address modes when entering server addresses.

DHCP Menu

The DHCP menu is accessible only when the DHCP client is enabled. The following DHCP configuration parameters can be modified on the DHCP menu:

Name	Possible Values	Description
Timeout	1 through 600	Number of seconds the phone waits for secondary DHCP Offer messages before selecting an offer.
Boot Server	0=Option 66	<p>The phone will look for option number 66 (string type) in the response received from the DHCP server. The DHCP server should send address information in option 66 that matches one of the formats described for Server Address in the following section, Server Menu.</p> <p>If the DHCP server sends nothing, the following scenarios are possible:</p> <ul style="list-style-type: none"> • If a boot server value is stored in flash memory and the value is not “0.0.0.0”, then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. <p>- If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value.</p> <p>- If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.</p>

Name	Possible Values	Description
Boot Server (continued)	1=Custom	<p>The phone will look for the option number specified by the Boot Server Option parameter (below), and the type specified by the Boot Server Option Type parameter (below) in the response received from the DHCP server.</p> <p>If the DHCP server sends nothing, the following scenarios are possible:</p> <ul style="list-style-type: none"> • If a boot server value is stored in flash memory and the value is not "0.0.0.0", then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. <p>- If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value.</p> <p>- If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.</p>
	2=Static	<p>The phone will use the boot server configured through the Server Menu. For more information, refer to the following section, Server Menu.</p>
	3=Custom+Option 66	<p>The phone will first use the custom option if present or use Option 66 if the custom option is not present.</p> <p>If the DHCP server sends nothing, the following scenarios are possible:</p> <ul style="list-style-type: none"> • If a boot server value is stored in flash memory and the value is not "0.0.0.0", then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. <p>- If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value. The phone prefers the custom option value over the Option 66 value, but if no custom option is given, the phone will use the Option 66 value.</p> <p>- If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.</p>
Boot Server Option	128 through 254 (Cannot be the same as VLAN ID Option)	<p>When the boot server parameter is set to Custom, this parameter specifies the DHCP option number in which the phone will look for its boot server.</p>
Boot Server Option Type	0=IP Address, 1=String	<p>When the Boot Server parameter is set to Custom, this parameter specifies the type of the DHCP option in which the phone will look for its boot server. The IP Address must specify the boot server. The String must match one of the formats described for Server Address in the following section, Server Menu.</p>

Name	Possible Values	Description
VLAN Discovery	0=Disabled (default)	No VLAN discovery through DHCP.
	1=Fixed	Use predefined DHCP vendor-specific option values of 128, 144, 157 and 191. If this is used, the VLAN ID Option field will be ignored
	2=Custom	Use the number specified in the VLAN ID Option field as the DHCP private option value.
VLAN ID Option	128 through 254 (Cannot be the same as Boot Server Option) (default is 129)	The DHCP private option value (when VLAN Discovery is set to Custom). For more information, refer to Assigning a VLAN ID Using DHCP on page C-16.

Note

If multiple alternate DHCP servers respond:

- The phone should gather the responses from alternate DHCP servers.
- If configured for Custom+Option66, the phone will select the first response that contains a valid "custom" option value.
- If none of the responses contain a "custom" option value, the phone will select the first response that contains a valid "option66" value.

Server Menu

The following server configuration parameters can be modified on the Server menu:

Name	Possible Values	Description
Server Type	0=FTP, 1=TFTP, 2=HTTP, 3=HTTPS, 4=FTPS, 5=Invalid	The protocol that the phone will use to obtain configuration and phone application files from the boot server. Refer to Supported Provisioning Protocols on page 3-4. Note: Active FTP is not supported for bootROM version 3.0 or later. Passive FTP is still supported.

Name	Possible Values	Description
Server Address	dotted-decimal IP address OR domain name string OR URL All addresses can be followed by an optional directory and optional file name.	The boot server to use if the DHCP client is disabled, the DHCP server does not send a boot server option, or the Boot Server parameter is set to Static. The phone can contact multiple IP addresses per DNS name. These redundant boot servers must all use the same protocol. If a URL is used it can include a user name and password. Refer to Supported Provisioning Protocols on page 3-4. A directory and the master configuration file can be specified. Note: ":", "@", or "/" can be used in the user name or password these characters if they are correctly escaped using the method specified in RFC 1738.
Server User	any string	The user name used when the phone logs into the server (if required) for the selected Server Type . Note: If the Server Address is a URL with a user name, this will be ignored.
Server Password	any string	The password used when the phone logs in to the server if required for the selected Server Type . Note: If the Server Address is a URL with user name and password, this will be ignored.
File Transmit Tries	1 to 10 Default 3	The number of attempts to transfer a file. (An attempt is defined as trying to download the file from all IP addresses that map to a particular domain name.)
Retry Wait	0 to 300 Default 1	The minimum amount of time that must elapse before retrying a file transfer, in seconds. The time is measured from the start of a transfer attempt which is defined as the set of upload/download transactions made with the IP addresses that map to a given boot server's DNS host name. If the set of transactions in an attempt is equal to or greater than the Retry Wait value, then there will be no further delay before the next attempt is started. For more information, refer to Deploying Phones From the Boot Server on page 3-15.
Provisioning Method	Default or SAS-VP	If SAS-VP is selected, provisioning is done (in addition to the normal process).
Network	Cable/DSL, LAN, Dial-up	The network environment the phone is operating in. The default value is Cable/DSL.
Tag SN to UA	Disabled, Enabled	If enabled, the phone's serial number (MAC address) is included in the User-Agent header of the Microbrowser. The default value is Disabled.
Provisioning String	any string	The URL used in XML post/response transactions. If empty, the configured URL is used. This field is disabled when Provisioning Method is Default.

Note

The **Server User** and **Server Password** parameters should be changed from the default values. Note that for insecure protocols the user chosen should have very few privileges on the server.

Ethernet Menu

The following Ethernet configuration parameters can be modified on the Ethernet menu:

Name	Possible Values	Description
CDP	Enabled, Disabled	If enabled, the phone will use CDP. It also reports PoE power usage to the switch. The default value is Enabled.
VLAN ID	Null, 0 through 4094	Phone's 802.1Q VLAN identifier. The default value is Null. Note: Null = no VLAN tagging
VLAN Filtering	Enabled, Disabled	Filter received Ethernet packets so that the TCP/IP stack does not process bad data or too much data. Enable/disable the VLAN filtering state. The default value is Enabled.
Storm Filtering	Enabled, Disabled	Filter received Ethernet packets so that the TCP/IP stack does not process bad data or too much data. Enable/disable the DoS storm prevention state. The default value is Enabled.
LAN Port Mode	0 = Auto 1 = 10HD 2 = 10FD 3 = 100HD 4 = 100FD 5 = 1000FD	The network speed over the Ethernet. The default value is Auto.
PC Port Mode	Auto, 10HD, 10FD, 100HD, 100FD, 1000FD	The network speed over the Ethernet. The default value is Auto.

Note

The **LAN Port Mode** and **PC Port Mode** parameters are only available on SoundPoint IP 330, 430, 550, 560, 601, and 650 phones. HD means half duplex and FD means full duplex.

It is recommended that you leave the **LAN** and **PC** parameters set to Auto.

Only the SoundPoint IP 560 supports the LAN Port Mode and PC Port Mode setting of 1000FD.

Syslog Menu

Syslog is a standard for forwarding log messages in an IP network. The term "syslog" is often used for both the actual syslog protocol, as well as the application or library sending syslog messages.

The syslog protocol is a very simplistic protocol: the syslog sender sends a small textual message (less than 1024 bytes) to the syslog receiver. The receiver is commonly called "syslogd", "syslog daemon" or "syslog server". Syslog messages can be sent through UDP, TCP, or TLS. The data is sent in cleartext.

Syslog is supported by a wide variety of devices and receivers. Because of this, syslog can be used to integrate log data from many different types of systems into a central repository.

The syslog protocol is defined in RFC 3164. For more information on syslog, go to <http://www.ietf.org/rfc/rfc3164.txt?number=3164>.

The following syslog configuration parameters can be modified on the Syslog menu:

Name	Possible Values	Description
Server Address	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.
Server Type	None=0, UDP=1, TCP=2, TLS=3	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
Facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3164. The default value is 16, which maps to "local 0".
Render Level	0 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on <code>log.render.level</code> and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page A-77 . Note: Use left and right arrow keys to change values.
Prepend MAC Address	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.

Setting Up the Boot Server

The boot server can be on the local LAN or anywhere on the Internet.

Multiple boot servers can be configured by having the boot server DNS name map to multiple IP addresses. The default number of boot servers is one and the maximum number is eight. The following protocols are supported for redundant boot servers: HTTPS, HTTP, and FTP. For more information on the protocol used on each platform, refer to [Supported Provisioning Protocols](#) on page 3-4.

All of the boot servers must be reachable by the same protocol and the content available on them must be identical. The parameters described in section [Server Menu](#) on page 3-9 can be used to configure the number of times each server will be tried for a file transfer and also how long to wait between each attempt. The maximum number of servers to be tried is configurable. For more information, contact your Certified Polycom Reseller.

Note

Be aware of how logs, overrides and directories are uploaded to servers that maps to multiple IP addresses. The server that these files are uploaded to may change over time.

If you want to use redundancy for uploads, synchronize the files between servers in the background.

However, you may want to disable the redundancy for uploads by specifying specific IP addresses instead of URLs for logs, overrides, and directory in the `<MAC-address>.cfg`.

To set up the boot server:**Note**

Use this procedure as a recommendation if this is your first boot server setup.

1. Install boot server application or locate suitable existing server(s).



Polycom recommends that you use RFC-compliant servers.

2. Create account and home directory.

Note

If the provisioning protocol requires an account name and password, the server account name and password must match those configured in the phones. Defaults are: provisioning protocol: FTP, name: PlcmSplp, password: PlcmSplp.

Each phone may open multiple connections to the server.

The phone will attempt to upload log files, a configuration override file, and a directory file to the server. This requires that the phone's account has

delete, write, and read permissions. The phone will still function without these permissions, but will not be able to upload files.

The files downloaded from the server by the phone should be made read-only.

Note

Typically all phones are configured with the same server account, but the server account provides a means of conveniently partitioning the configuration. Give each account an unique home directory on the server and change the configuration on an account-by-account basis.

3. Copy all files from the distribution zip file to the phone home directory. Maintain the same folder hierarchy.

The distribution zip file contains:

- **sip.ld** (including a separate one for every supported model)
- **sip.cfg**
- **phone1.cfg**
- **000000000000.cfg**
- **000000000000-directory~.xml**
- **SoundPointIP-dictionary.xml** (one of each supported language)
- **SoundPointIPWelcome.wav**

Refer to the *Release Notes* for a detailed description of each file in the distribution.

Boot Server Security Policy

You must decide on a boot server security policy.



Polycom recommends allowing file uploads to the boot server where the security environment permits. This allows event log files to be uploaded and changes made by the phone user to the configuration (through the web server and local user interface) and changes made to the directory to be backed up.

For organizational purposes, configuring a separate log file directory is recommended, but not required. (For more information on LOG_FILE_DIRECTORY, refer to [Master Configuration Files](#) on page A-2.)

File permissions should give the minimum access required and the account used should have no other rights on the server.

The phone's server account needs to be able to add files to which it can write in the log file directory and the root directory. It must also be able to list files in all directories mentioned in the **<MAC-address>.cfg** file. All other files that

the phone needs to read, such as the application executable and the standard configuration files, should be made read-only through file server file permissions.

Deploying Phones From the Boot Server

You can successfully deploy SoundPoint IP and SoundStation IP phones from one or more boot servers.

Multiple boot servers can be configured by having the boot server DNS name map to multiple IP addresses. The default number of boot servers is one and the maximum number is eight. HTTPS, HTTP, and FTP are supported for redundant boot servers.

To deploy phones from the boot server:

Note

For more information on encrypting configuration files, refer to [Encrypting Configuration Files](#) on page C-3.

1. (Optional) Create per-phone configuration files by performing the following steps:

Note

This step may be omitted if per-phone configuration is not needed.

- a Obtain a list of phone Ethernet addresses (barcoded label on underside of phone and on the outside of the box).
- b Create per-phone **phone[MACaddress].cfg** file by using the **phone1.cfg** file from the distribution as templates.

For more information on the **phone1.cfg** file, refer to [Per-Phone Configuration](#) on page A-89.

Note

Throughout this guide, the terms Ethernet address and MAC address are used interchangeably.

- c Edit contents of **phone[MACaddress].cfg** if desired.

For example, edit the parameters.

2. (Optional) Create new configuration file(s) in the style of **sip.cfg** by performing the following steps:

Note

For more information on why to create another configuration file, refer to the "Configuration File Management on SoundPoint IP Phones" whitepaper at www.polycom.com/support/voice/.

For more information, especially on the SIP server address, refer to [SIP <SIP/>](#) on page A-10.

For more information on the **sip.cfg** file, refer to [Application Configuration](#) on page A-4.

Most of the default settings are typically adequate, however, if SNTP settings are not available through DHCP, the SNTP GMT offset and (possibly) the SNTP server address will need to be edited for the correct local conditions. Changing the default daylight savings parameters will likely be necessary outside of North American locations.

- a (Optional) Disable the local web (HTTP) server or change its signalling port if local security policy dictates.
 - b Change the default location settings for user interface language and time and date format.
3. (Optional) Create a master configuration file by performing the following steps:
 - a Create per-phone or per-platform **<Ethernet address>.cfg** files by using the **000000000000.cfg** and files from the distribution as templates.

For more information, refer to [Master Configuration Files](#) on page A-2.

- b Edit the CONFIG_FILES attribute of the **<Ethernet address>.cfg** files so that it references the appropriate **phone[MACaddress].cfg** file.

For example, replace the reference to **phone1.cfg** with **phone[MACaddress].cfg**.

- c Edit the CONFIG_FILES attribute of the **<Ethernet address>.cfg** files so that it references the appropriate **sipXXXX.cfg** file.

For example, replace the reference to **sip.cfg** with **sip650.cfg**.

- d Edit the LOG_FILE_DIRECTORY attribute of the **<Ethernet address>.cfg** files so that it points to the log file directory.
- e Edit the CONTACT_DIRECTORY attribute of the **<Ethernet address>.cfg** files so that it points to the organization's contact directory.

4. Reboot the phones by pressing the reboot multiple key combination.

For more information, refer to [Multiple Key Combinations](#) on page C-9.

The bootROM and SIP application modify the APPLICATION APP_FILE_PATH attribute of the <Ethernet address>.cfg files so that it references the appropriate sip.ld files.

For example, the reference to sip.ld is changed to 2345-11605-001.sip.ld to boot the SoundPoint IP 601 image.

Note

At this point, the phone sends a DHCP Discover packet to the DHCP server. This is found in the Bootstrap Protocol/option "Vendor Class Identifier" section of the packet and includes the phone's part number and the bootROM version.

For example, a SoundPoint IP 650 might send the following information:

```
5EL@
DC?5cSc52*46*(9N7*<u6=pPolycomSoundPointIP-SPIP_6502345-12600-001,1B
R/4.0.0.0155/23-May-07 13:35BR/4.0.0.0155/23-May-07 13:35
```

For more information, refer to [Parsing Vendor ID Information](#) on page C-17.

5. Monitor the boot server event log and the uploaded event log files (if permitted).

Ensure that the configuration process completed correctly. All configuration files used by the boot server are logged.

You can now instruct your users to start making calls.

Upgrading SIP Application

You can upgrade the SIP application that is running on the SoundPoint IP and SoundStation IP phones in your organization. The exact steps that you perform are dependent on the version of the SIP application that is currently running on the phones and the version that want to upgrade to.

The bootROM, application executable, and configuration files can be updated automatically through the centralized provisioning model. These files are read-only by default.

Most organization can use the instructions shown in the next section, [Supporting SoundPoint IP and SoundStation IP Phones](#).

However, if your organization has a mixture of SoundPoint IP 300 and/or 500 phones deployed along with other models, you will need to change the phone configuration files to continue to support the SoundPoint IP 300 and IP 500 phones when software releases SIP 2.2.0 or later are deployed. These models were discontinued as of May 2006. In this case, refer to [Supporting SoundPoint IP 300 and 500 Phones](#) on page 3-19.

Warning

The SoundPoint IP 300 and 500 phones will be supported on the latest maintenance patch release of the SIP 2.1 software stream—currently SIP 2.1.2. Any critical issues that affect SoundPoint IP 300 and 500 phones will be addressed by a maintenance patch on this stream until the End of Life date for these products. Phones should be upgraded to BootROM 4.0.0 for these changes to be effective.

Supporting SoundPoint IP and SoundStation IP Phones

To automatically update:

1. Back up old application and configuration files.

The old configuration can be easily restored by reverting to the backup files.

2. Customize new configuration files or apply new or changed parameters to the old configuration files.

Differences between old and new versions of configuration files are explained in the *Release Notes* that accompany the software. Both mandatory and optional changes may present. Changes to site-wide configuration files such as **sip.cfg** can be done manually, but a scripting tool is useful to change per-phone configuration files.

Warning

The configuration files listed in CONFIG_FILES attribute of the master configuration file must be updated when the software is updated. Any new configuration files must be added to the CONFIG_FILES attribute in the appropriate order.

Mandatory changes must be made or the software may not behave as expected.

For more information, refer to the “Configuration File Management on SoundPoint IP Phones” whitepaper at www.polycom.com/support/voice/.

3. Save the new configuration files and images (such as **sip.ld**) on the boot server.
4. Reboot the phones by pressing the reboot multiple key combination.

For more information, refer to [Multiple Key Combinations](#) on page C-9.

Since the APPLICATION APP_FILE_PATH attribute of the **<Ethernet address>.cfg** files references the individual **sip.ld** files, it is possible to verify that an update is applied to phones of a particular model.

For example, the reference to **sip.ld** is changed to **2345-11605-001.sip.ld** to boot the SoundPoint IP 601 image.

The phones can be rebooted remotely through the SIP signaling protocol. Refer to [Special Events <specialEvent/>](#) on page A-15.

The phones can be configured to periodically poll the boot server to check for changed configuration files or application executable. If a change is detected, the phone will reboot to download the change. Refer to [Provisioning <prov/>](#) on page [A-81](#).

Supporting SoundPoint IP 300 and 500 Phones

With enhancements in BootROM 4.0.0 and SIP 2.1.2, you can modify the `000000000000.cfg` or `<Ethernet address>.cfg` configuration file to direct phones to load the software image and configuration files based on the phone model number. Refer to [Master Configuration Files](#) on page [A-2](#).

The SIP 2.2.0 or later software distributions contain only the new distribution files for the new release. You must rename the `sip.ld`, `sip.cfg`, and `phone1.cfg` from a previous 2.1.2 distribution that is compatible with SoundPoint IP 300 and 500 phones.

The following procedure must be used for upgrading to SIP 2.2.0 or later for installations that have SoundPoint IP 300 and 500 phones deployed. It is also recommended that this same approach be followed even if SoundPoint IP 300 and 500 phones are not part of the deployment as it will simplify management of phone systems with future software releases.

To upgrade your SIP application:

1. Do one of the following steps:
 - a Place the `bootrom.ld` file corresponding to BootROM revision 4.0.0 (or later) onto the boot server.
 - b Ensure that all phones are running BootROM 4.0.0 or later code.
2. Copy `sip.ld`, `sip.cfg` and `phone1.cfg` from the SIP2.2.0 or later release distribution onto the boot server.

These are the relevant files for all phones except the SoundPoint IP 300 and 500 phones.

3. Rename `sip.ld`, `sip.cfg`, and `phone1.cfg` from the previous distribution to `sip_212.ld`, `sip_212.cfg`, and `phone1_212.cfg` respectively on the boot server.

These are the relevant files for supporting the SoundPoint IP 300 and 500 phones.

4. Modify the `000000000000.cfg` file, if required, to match your configuration file structure.

For example:

```
<APPLICATION
APP_FILE_PATH="sip.ld"
APP_FILE_PATH_SPIP500="sip_212.ld"
APP_FILE_PATH_SPIP300="sip_212.ld"
```

```
CONFIG_FILES="[PHONE_MAC_ADDRESS]-user.cfg, phone1.cfg, sip.cfg"
CONFIG_FILES_SPIP500="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_212.cfg, sip_212.cfg"
CONFIG_FILES_SPIP300="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_212.cfg, sip_212.cfg"
MISC_FILES=""
LOG_FILE_DIRECTORY=""
OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY=""
/>
```

5. Remove any **<Ethernet address>.cfg** files that may have been used with earlier releases from the boot server.

Note

This approach takes advantage of an enhancement that was added in SIP2.0.1/BootROM 3.2.1 that allows for the substitution of the phone specific [MACADDRESS] inside configuration files. This avoids the need to create unique **<Ethernet address>.cfg** files for each phone such that the default **000000000000.cfg** file can be used for all phones in a deployment.

If this approach is not used, then changes will need to be made to all the **<Ethernet address>.cfg** files for SoundPoint IP 300 and 500 phones or all of the **<Ethernet address>.cfg** files if it is not explicitly known which phones are SoundPoint IP 300 and 500 phones.

For more information, refer to “Technical Bulletin 35311: Supporting SoundPoint IP 300 and IP 500 Phones with SIP 2.2 and Later Releases” at <http://www.polycom.com/support/voice/>.

Configuring Your System

After you set up your SoundPoint IP / SoundStation IP phones on the network, you can allow users to place and answer calls using the default configuration, however, you may be require some basic changes to optimize your system for best results.

This chapter provides information for making configuration changes for:

- [Setting Up Basic Features](#)
- [Setting Up Advanced Features](#)
- [Setting Up Audio Features](#)
- [Setting Up Security Features](#)

This chapter also provides instructions on:

- [Configuring SoundPoint IP / SoundStation IP Phones Locally](#)

To troubleshoot any problems with your SoundPoint IP / SoundStation IP phones on the network, refer to [Troubleshooting Your SoundPoint IP / SoundStation IP Phones](#) on page 5-1. For more information on the configuration files, refer to [Configuration Files](#) on page A-1.

Setting Up Basic Features

This section provides information for making configuration changes for the following basic features:

- [Call Log](#)
- [Call Timer](#)
- [Call Waiting](#)
- [Called Party Identification](#)
- [Calling Party Identification](#)
- [Missed Call Notification](#)

- [Connected Party Identification](#)
- [Context Sensitive Volume Control](#)
- [Customizable Audio Sound Effects](#)
- [Message Waiting Indication](#)
- [Distinctive Incoming Call Treatment](#)
- [Distinctive Ringing](#)
- [Distinctive Call Waiting](#)
- [Do Not Disturb](#)
- [Handset, Headset, and Speakerphone](#)
- [Local Contact Directory](#)
- [Local Digit Map](#)
- [Microphone Mute](#)
- [Soft Key Activated User Interface](#)
- [Speed Dial](#)
- [Time and Date Display](#)
- [Idle Display Animation](#)
- [Ethernet Switch](#)
- [Graphic Display Backgrounds](#)

This section also provides information for making configuration changes for the following basic call management features:



- [Automatic Off-Hook Call Placement](#)
- [Call Hold](#)
- [Call Transfer](#)
- [Local / Centralized Conferencing](#)
- [Call Forward](#)
- [Directed Call Pick-Up](#)
- [Group Call Pick-Up](#)
- [Call Park/Retrieve](#)
- [Last Call Return](#)

Call Log

The phone maintains a call log. The log contains call information such as remote party identification, time and date, and call duration. It can be used to redial previous outgoing calls, return incoming calls, and save contact information from call log entries to the contact directory.

The call log is stored in volatile memory and is maintained automatically by the phone in three separate lists: Missed Calls, Received Calls and Placed Calls. The call lists can be cleared manually by the user and will be erased when the phone is restarted.

Note

On some SoundPoint IP platforms, missed calls and received calls appear in one list. Missed calls appear as  and received calls appear as .

The “call list” feature can be disabled on all SoundPoint IP and SoundStation IP platforms except the SoundPoint IP 330/320.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: sip.cfg	Enable or disable all call lists or individual call lists. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
----------------------------------	---------------------------------------	--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

Call Timer

A call timer is provided on the display. A separate call timer is maintained for each distinct call in progress. The call duration appears in hours, minutes, and seconds.

There are no related configuration changes.

Call Waiting

When an incoming call arrives while the user is active on another call, the incoming call is presented to the user visually on the LCD display. A configurable sound effect such as the familiar call-waiting beep will be mixed with the active call audio as well.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: phone1.cfg	Specify the ring tone heard on an incoming call when another call is active. <ul style="list-style-type: none"> For more information, refer to Call Waiting <callWaiting/> on page A-95.
----------------------------------	------------------------------------------	-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

For related configuration changes, refer to [Customizable Audio Sound Effects](#) on page [4-5](#).

Called Party Identification

The phone displays and logs the identity of the remote party specified for outgoing calls. This is the party that the user intends to connect with.

There are no related configuration changes.

Calling Party Identification

The phone displays the caller identity, derived from the network signalling, when an incoming call is presented, if the information is provided by the call server. For calls from parties for which a directory entry exists, the local name assigned to the directory entry may optionally be substituted.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration File: sip.cfg	Specify whether or not to use directory name substitution. <ul style="list-style-type: none"> For more information, refer to User Preferences <up/> on page A-23.
Local	Web Server (if enabled)	Specify whether or not to use directory name substitution. Navigate to: <code>http://<phoneIPAddress>/coreConf.htm#us</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Missed Call Notification

The phone can display the number of calls missed since the user last looked at the Missed Calls list. The types of calls that are counted as “missed” can be configured per registration. Remote missed call notification can be used to notify the phone when a call originally destined for it is diverted by another entity such as a Session Initiation Protocol (SIP) server.

Note

On some SoundPoint IP platforms, missed calls and received calls appear in one list.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
	Configuration file: phone1.cfg	Specify per-registration whether all missed-call events or only remote/server-generated missed-call events will be displayed. <ul style="list-style-type: none"> For more information, refer to Missed Call Configuration <serverMissedCall/> on page A-94.

Connected Party Identification

The identity of the remote party to which the user has connected is displayed and logged, if the name and ID is provided by the call server. The connected party identity is derived from the network signaling. In some cases the remote party will be different from the called party identity due to network call diversion.

There are no related configuration changes.

Context Sensitive Volume Control

The volume of user interface sound effects, such as the ringer, and the receive volume of call audio is adjustable. While transmit levels are fixed according to the TIA/EIA-810-A standard, receive volume is adjustable. For SoundPoint IP and phones, if using the default configuration parameters, the receive handset/headset volume resets to nominal after each call to comply with regulatory requirements. Handsfree volume persists with subsequent calls.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Adjust receive and handset/headset volume. <ul style="list-style-type: none"> For more information, refer to Volume Persistence <volume/> on page A-37.
----------------------------------	---------------------------------------	--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

Customizable Audio Sound Effects

Audio sound effects used for incoming call alerting and other indications are customizable. Sound effects can be composed of patterns of synthesized tones or sample audio files. The default sample audio files may be replaced with alternates in **.wav** file format. Supported **.wav** formats include:

- mono G.711 (13-bit dynamic range, 8-khz sample rate)
- mono L16/16000 (16-bit dynamic range, 16-kHz sample rate)

Note

L16/16000 is not supported on SoundPoint IP 301 and SoundStation IP 4000 phones.

Note

The alternate sampled audio sound effect files must be present on the boot server or the Internet for downloading at boot time.

Configuration changes can performed centrally at the boot server or locally:

Central (boot server)	Configuration File: sip.cfg	Specify patterns used for sound effects and the individual tones or sampled audio files used within them. <ul style="list-style-type: none"> For more information, refer to Sampled Audio for Sound Effects <saf/> on page A-27 or Sound Effects <se/> on page A-28.
Local	Web Server (if enabled)	Specify sampled audio wave files to replace the built-in defaults. Navigate to <code>http://<phoneIPAddress>/coreConf.htm#sa</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Message Waiting Indication

The phone will flash a message-waiting indicator (MWI) LED when instant messages and voice messages are waiting.

Configuration changes can performed centrally at the boot server:

Central (boot server)	Configuration file: phone1.cfg	Specify per-registration whether the MWI LED is enabled or disabled. <ul style="list-style-type: none"> For more information, refer to Message Waiting Indicator <mw/> on page A-102. Specify whether MWI notification is displayed for registration x (pre-SIP 2.1 behavior is enabled). <ul style="list-style-type: none"> For more information, refer to User Preferences <up/> on page A-23.
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Distinctive Incoming Call Treatment

The phone can automatically apply distinctive treatment to calls containing specific attributes. The distinctive treatment that can be applied includes customizable alerting sound effects and automatic call diversion or rejection. Call attributes that can trigger distinctive treatment include the calling party name or SIP contact (number or URL format).

For related configuration changes, refer to [Local Contact Directory](#) on page 4-9.

Distinctive Ringing

There are three options for distinctive ringing:

1. The user can select the ring type for each line. This option has the lowest priority.
2. The ring type for specific callers can be assigned in the contact directory. For more information, refer to [Distinctive Incoming Call Treatment](#), the previous section. This option has a higher priority than option 1 and a lower priority than option 3.
3. The `voIpProt.SIP.alertInfo.x.value` and `voIpProt.SIP.alertInfo.x.class` fields can be used to map calls to specific ring types. This option has the highest priority.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the mapping of Alert-Info strings to ring types. <ul style="list-style-type: none"> • For more information, refer to Alert Information <alertInfo/> on page A-14.
	Configuration file: phone1.cfg	Specify the ring type to be used for each line. <ul style="list-style-type: none"> • For more information, refer to Registration <reg/> on page A-89.
	XML File: <Ethernet address>-directory.xml	This file can be created manually using an XML editor. <ul style="list-style-type: none"> • For more information, refer to Local Contact Directory on page 4-9.
Local	Local Phone User Interface	The user can edit the ring types selected for each line under the Settings menu. The user can also edit the directory contents. Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Distinctive Call Waiting

The `voIpProt.SIP.alertInfo.x.value` and `voIpProt.SIP.alertInfo.x.class` fields can be used to map calls to distinct call waiting types, currently limited to two styles.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify the mapping of Alert-Info strings to call waiting types. <ul style="list-style-type: none"> • For more information, refer to Alert Information <alertInfo/> on page A-14.
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Do Not Disturb

A Do Not Disturb (DND) feature is available to temporarily stop all incoming call alerting. Calls can optionally be treated as though the phone is busy while DND is enabled. DND can be configured as a per-registration feature. Incoming calls received while DND is enabled are logged as missed. For more information on forwarding calls while DND is enabled, refer to [Call Forward](#) on page 4-20.

Server-based DND is active if the feature is enabled on both the phone and the server and the phone is registered. The server-based DND feature is applicable for all registrations on the phone (no per-registration mode) and it disables local Call Forward and DND features.

Server-based DND will behave the same as per-SIP2.1 per-registration feature with the following exceptions:

- There is no indication on the phone's user interface whether or not server-based DND is active.
- If server-based DND is enabled, but inactive, and the user presses the DND key or selects the DND option on the Feature menu, the "Do Not Disturb" message does not appear on the user's phone (incoming call alerting will continue).

Note

Server-based DND is disabled if Shared Call Appearance or Bridged Line Appearance is enabled.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Enable or disable server-based DND. <ul style="list-style-type: none"> • For more information, refer to SIP <SIP/> on page A-10 Specify whether or not DND results in incoming calls being given busy treatment. <ul style="list-style-type: none"> • For more information, refer to Call Handling Configuration <call/> on page A-58.
	Configuration file: phone1.cfg	Enable or disable server-based DND as a per-registration feature. <ul style="list-style-type: none"> • For more information, refer to Registration <reg/> on page A-89. Specify whether DND is treated as a per-registration feature or a global feature on the phone. <ul style="list-style-type: none"> • For more information, refer to Do Not Disturb <dnd/> on page A-98.
Local	Local Phone User Interface	Enable or disable DND using the "Do Not Disturb" key on the SoundPoint IP 301, 501, 550, 560, 600, 601, and 650 or the "Do Not Disturb" option on the Features menu on the SoundPoint IP 320, 330, and 430 and SoundStation IP 4000.

Handset, Headset, and Speakerphone

SoundPoint IP phones come standard with a handset and a dedicated connector is provided for a headset (not supplied). The SoundPoint IP 320, 330, 430, 500, 501, 550, 560, 600, 601, and 650 desktop phones and SoundStation IP 4000 conference phone are full-duplex speakerphones. The SoundPoint IP 301 phones is a listen-only speakerphone. The SoundPoint IP phones provide dedicated keys for convenient selection of either the speakerphone or headset.

Only the SoundPoint IP 320, 330, 430, 550, 560, 600, 601, and 650 desktop phones can be configured to use the electronic hookswitch. For more information, refer to “Technical Bulletin 35150: Using an Electronic Hookswitch with SoundPoint IP Phones” at <http://www.polycom.com/support/voip/>.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Enable or disable persistent headset mode. For more information, refer to User Preferences <up/> on page A-23. Enable or disable hands-free speakerphone mode. <ul style="list-style-type: none"> For more information, refer to User Preferences <up/> on page A-23.
	Configuration file: phone1.cfg	Specify whether or not the electronic hookswitch is enabled and what type of headset is attached. <ul style="list-style-type: none"> For more information, refer to User Preferences <user_preferences/> on page A-89.
Local	Web Server (if enabled)	Enable or disable persistent headset mode. Navigate to: <a href="http://<phoneIPAddress>/coreConf.htm#us">http://<phoneIPAddress>/coreConf.htm#us
	Local Phone User Interface	Enable or disable persistent headset mode through the Settings menu (Settings > Basic > Preferences > Headset > Headset Memory Mode). Enable or disable hands-free speakerphone mode through the Settings menu (Settings > Advanced > Admin Settings > Phone Settings). Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Local Contact Directory

The phone maintains a local contact directory. The directory can be downloaded from the boot server and edited locally (if configured in that way). Contact information from previous calls may be easily added to the directory for convenient future access.

The directory is the central database for several other features including speed-dial, distinctive incoming call treatment, presence, and instant messaging.

Note

If a user makes a change to the local contact directory, there is a five second timeout before it is uploaded to the boot server as **<mac-address>-directory.cfg**.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Set whether the directory uses volatile storage on the phone (required on the SoundPoint IP 500 platform for directories greater than 25 entries). <ul style="list-style-type: none"> For more information, refer to Local Directory <local/> on page A-62. Specify whether or not the local contact directory is read only. <ul style="list-style-type: none"> For more information, refer to Local Directory <local/> on page A-62.
	XML file: 00000000000-direct ory.xml	A sample file named 00000000000-directory~.xml (Note the extra “~” in the filename) is included with the application file distribution. This file can be used as a template for the per-phone <Ethernet address>-directory.xml directories (edit contents, then rename to <Ethernet address>-directory.xml). It also can be used to seed new phones with an initial directory (edit contents, then remove “~” from file name). Telephones without a local directory, such as new units from the factory, will download the 00000000000-directory.xml directory and base their initial directory on it. These files should be edited with an XML editor. These files can be downloaded once per reflash. For information on file format, refer to Local Contact Directory File Format , the following section.
	XML file: <Ethernet address>-directory.xml	This file can be created manually using an XML editor. For information on file format, refer to Local Contact Directory File Format , the following section.
Local	Local Phone User Interface	The user can edit the directory contents if configured in that way. Changes will be stored in the phone’s flash file system and backed up to the boot server copy of <Ethernet address>-directory.xml if this is configured. When the phone boots, the boot server copy of the directory, if present, will overwrite the local copy.

Local Contact Directory File Format

An example of a local contact directory is shown below. The subsequent table provides an explanation of each element.

```
<?xml version="1.0" encoding="UTF-8" standalone="yes" ?>
<directory>
  <item_list>
```

```

<item>
  <ln>Doe</ln>
  <fn>John</fn>
  <ct>1001</ct>
  <sd>1</sd>
  <rt>1</rt>
  <dc/>
  <ad>0</ad>
  <ar>0</ar>
  <bw> 0</bw>
  <bb>0</bb>
</item>
...
<item>
  <ln>Smith</ln>
  <fn>Bill</fn>
  <ct>1003</ct>
  <sd>3</sd>
  <rt>3</rt>
  <dc/>
  <ad>0</ad>
  <ar>0</ar>
  <bw> 0</bw>
  <bb>0</bb>
</item>
</item_list>
</directory>

```

Element	Permitted Values	Interpretation
fn	UTF-8 encoded string of up to 40 bytes	first name Note: In some cases, this will be less than 40 characters due to UTF-8's variable length encoding.
ln	UTF-8 encoded string of up to 40 bytes	last name
ct	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL	contact Used by the phone to address a remote party in the same way that a string of digits or a SIP URL are dialed manually by the user. This element is also used to associate incoming callers with a particular directory entry. Note: This field cannot be null or duplicated.
sd	Null, 1 to 9999	speed-dial index Associates a particular entry with a speed dial bin for one-touch dialing or dialing from the speed dial menu. Note: On the SoundPoint IP 330/320, the maximum speed-dial index is 99.

Element	Permitted Values	Interpretation
rt	Null, 1 to 21	ring type When incoming calls can be associated with a directory entry by matching the address fields, this field is used to specify ring type to be used.
dc	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL	divert contact The forward-to address for the autodivert feature.
ad	0,1	auto divert If set to 1, automatically diverts callers that match the directory entry to the address specified in divertcontact. Note: <i>If auto-divert is enabled, it has precedence over auto-reject.</i>
ar	0,1	auto-reject If set to 1, automatically rejects callers that match the directory entry. Note: <i>If auto-divert is also enabled, it has precedence over auto-reject.</i>
bw	0,1	buddy watching If set to 1, add this contact to the list of watched phones.
bb	0,1	buddy block If set to 1, block this contact from watching this phone.

Local Digit Map

The phone has a local digit map feature to automate the setup phase of number-only calls. When properly configured, this feature eliminates the need for using the **Dial** or **Send** soft key when making outgoing calls. As soon as a digit pattern matching the digit map is found, the call setup process will complete automatically. The configuration syntax is based on recommendations in 2.1.5 of RFC 3435. The phone behavior when the user dials digits that do not match the digit map is configurable. It is also possible to strip a trailing # from the digits sent or to replace certain matched digits (with the introduction of "R" to the digit map).

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value. <ul style="list-style-type: none"> For more information, refer to Dial Plan <dialplan/> on page A-16.
	Configuration file: phone1.cfg	Specify per-registration impossible match behavior, trailing # behavior, digit map matching strings, and time out values that override those in sip.cfg . <ul style="list-style-type: none"> For more information, refer to Dial Plan <dialplan/> on page A-98.
Local	Web Server (if enabled)	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value. Navigate to: <code>http://<phoneIPAddress>/appConf.htm#ls</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Microphone Mute

A microphone mute feature is provided. When activated, visual feedback is provided. This is a local function and cannot be overridden by the network.

There are no related configuration changes.

Soft Key Activated User Interface

The user interface makes extensive use of intuitive, context-sensitive soft key menus. The soft key function is shown above the key on the graphic display.

There are no related configuration changes.

Speed Dial

Entries in the local directory can be linked to the speed dial system. The speed dial system allows calls to be placed quickly from dedicated keys as well as from a speed dial menu.

If Presence watching is enabled for speed dial entries, their status will be shown on the idle display (if the SIP server supports this feature). For more information, refer to [Presence](#) on page 4-43.

Configuration changes can be performed centrally at the boot server or locally:

<p>Central (boot server)</p>	<p>XML file: <Ethernet address>-directory. xml</p>	<p>The <code><sd>x</sd></code> element in the <Ethernet address>-directory.xml file links a directory entry to a speed dial resource within the phone. Speed dial entries are mapped automatically to unused line keys (line keys are not available on the SoundStation IP 4000) and are available for selection within the speed dial menu. (Press the up-arrow key from the idle display to jump to SpeedDial).</p> <ul style="list-style-type: none"> For more information, refer to Local Contact Directory on page 4-9.
<p>Local</p>	<p>Local Phone User Interface</p>	<p>The next available Speed Dial Index is assigned to new directory entries. Key pad short cuts are available to facilitate assigning and modifying the Speed Dial Index value for entries in the directory. The Speed Dial Index field is used to link directory entries to speed dial operations.</p> <p>Changes will be stored in the phone's flash file system and backed up to the boot server copy of <Ethernet address>-directory.xml if this is configured. When the phone boots, the boot server copy of the directory, if present, will overwrite the local copy.</p>

Time and Date Display

The phone maintains a local clock and calendar. Time and date can be displayed in certain operating modes such as when the phone is idle and during a call. The clock and calendar must be synchronized to a remote Simple Network Time Protocol (SNTP) timeserver. The time and date displayed on the phone will flash continuously until a successful SNTP response is received to indicate that they are not accurate. The time and date display can use one of several different formats and can be turned off.

Configuration changes can be performed centrally at the boot server or locally:

<p>Central (boot server)</p>	<p>Configuration file: sip.cfg</p>	<p>Turn time and date display on or off.</p> <ul style="list-style-type: none"> For more information, refer to User Preferences <up/> on page A-23. <p>Set the time and date display formats.</p> <ul style="list-style-type: none"> For more information, refer to Date and Time <datetime/> on page A-23. <p>Set the basic SNTP settings and daylight savings parameters.</p> <ul style="list-style-type: none"> For more information, refer to Time Synchronization <sntp/> on page A-53.
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Local	Web Server (if enabled)	<p>Set the basic SNTP and daylight savings settings.</p> <p>Navigate to: <code>http://<phoneIPAddress>/coreConf.htm#ti</code></p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.</p>
	Local Phone User Interface	<p>The basic SNTP settings can be made in the Network Configuration menu.</p> <p>For more information, refer to DHCP or Manual TCP/IP Setup on page 3-2.</p> <p>The user can edit the time and date format and enable or disable the time and date display under the Settings menu.</p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. They will permanently override global settings unless deleted through the Reset Local Config menu selection.</p>

Idle Display Animation

All phones except the SoundPoint IP 301 can display a customized animation on the idle display in addition to the time and date. For example, a company logo could be displayed (refer to [Adding a Background Logo](#) on page C-5).

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	<p>To turn idle display animation on or off.</p> <ul style="list-style-type: none"> For more information, refer to Indicators <ind/> on page A-71. <p>To replace the animation used for the idle display.</p> <ul style="list-style-type: none"> For more information, refer to Animations <anim/> <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/> on page A-72. <p>To change the position of the idle display animation.</p> <ul style="list-style-type: none"> For more information, refer to Graphic Icons <gi/> <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/> on page A-74.
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Ethernet Switch

The SoundPoint IP phones contain two Ethernet ports, labeled LAN and PC, and an embedded Ethernet switch that runs at full line-rate. The SoundStation IP phones contain only one Ethernet port, labeled LAN. The Ethernet switch allows a personal computer and other Ethernet devices to connect to the office LAN by daisy chaining through the phone, eliminating the need for a stand-alone hub. The SoundPoint IP switch gives higher transmit priority to packets originating in the phone. The phone can be powered through a local

AC power adapter or can be line-powered (power supplied through the signaling or idle pairs of the LAN Ethernet cable). Line powering typically requires that the phone plugs directly into a dedicated LAN jack. Devices that do not require LAN power can then plug into the SoundPoint IP PC Ethernet port.

SoundPoint IP Switch - Port Priorities

To help ensure good voice quality, the Ethernet switch embedded in the SoundPoint IP phones should be configured to give voice traffic emanating from the phone higher transmit priority than those from a device connected to the PC port. If not using a VLAN (VLAN set to blank in the setup menu), this will automatically be the case. If using a VLAN, ensure that the 802.1p priorities for both default and real-time transport protocol (RTP) packet types are set to 2 or greater. Otherwise, these packets will compete equally with those from the PC port. For more information, refer to [Quality of Service <QOS/>](#) on page [A-50](#).

Graphic Display Backgrounds

You can set up a picture or design to be displayed on the background of the graphic display of all SoundPoint IP 550, 560, and 650 phones. There are a number of default backgrounds, both solid and bitmaps. Users can select which background appear on their phone.

You can modify the solid backgrounds and bitmaps. For example, you can add a grey solid background or modify a bitmap to one of your choice.

Note

When installing a background of your choice, care needs to be taken to ensure that the background does not adversely affect the visibility of the text on the phone display. As a general rule, backgrounds should be light in shading for better usability.

To modify the backgrounds displayed on the supported SoundPoint IP phones:

1. Modify the `sip.cfg` configuration file as follows:
 - a. Open `sip.cfg` in an XML editor.
 - b. Locate the background parameter.
 - c. For the solid backgrounds, set the name and RGB values. For example:

```
bg.gf.gray.pat.solid.3.name="Gray"  
bg.gf.gray.pat.solid.3.red="128"  
bg.gf.gray.pat.solid.3.green="128"  
bg.gf.gray.pat.solid.3.blue="128"
```


- d** For the bitmaps, enter a bitmap filename and put the file on the boot server. For example:

```
bg.gf.gray.3.name="polycom.bmp"
```

The default size for bitmaps is 320 x 160. Smaller bitmaps will be centered and a white border added. Larger bitmaps will be centered and cropped to fit.

- e** Save the modified **sip.cfg** configuration file.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: phone1.cfg	Specify which backgrounds will be displayed. <ul style="list-style-type: none"> For more information, refer to Backgrounds <bg/> on page A-70.
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Automatic Off-Hook Call Placement

The phone supports an optional automatic off-hook call placement feature for each registration.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: phone1.cfg	Specify which registrations have the feature and what contact to call when going off hook. <ul style="list-style-type: none"> For more information, refer to Automatic Off-Hook Call Placement <autoOffHook/> on page A-94.
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Call Hold

The purpose of hold is to pause activity on one call so that the user may use the phone for another task, such as to make or receive another call. Network signaling is employed to request that the remote party stop sending media and to inform them that they are being held. A configurable local hold reminder feature can be used to remind the user that they have placed calls on hold. The call hold reminder is always played through the speakerphone.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify whether RFC 2543 (c=0.0.0.0) or RFC 3264 (a=sendonly or a=inactive) outgoing hold signaling is used. <ul style="list-style-type: none"> For more information, refer to SIP <SIP/> on page A-10. Specify local hold reminder options. <ul style="list-style-type: none"> For more information, refer to Hold, Local Reminder <hold/><localReminder/> on page A-61.
Local	Web Server (if enabled)	Specify whether or not to use RFC 2543 (c=0.0.0.0) outgoing hold signaling. The alternative is RFC 3264 (a=sendonly or a=inactive). Navigate to: <a href="http://<phoneIPAddress>/appConf.htm#s">http://<phoneIPAddress>/appConf.htm#s Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.
	Local Phone User Interface	Use the SIP Configuration menu to specify whether or not to use RFC 2543 (c=0.0.0.0) outgoing hold signaling. The alternative is RFC 3264 (a=sendonly or a=inactive).

Call Transfer

Call transfer enables the user (party A) to move an existing call (party B) into a new call between party B and another user (party C) selected by party A. The phone offers three types of transfers:

- Blind transfers – The call is transferred immediately to party C after party A has finished dialing party C's number. Party A does not hear ring-back.
- Attended transfers – Party A dials party C's number and hears ring-back and decides to complete the transfer before party C answers. This option can be disabled.
- Consultative transfers – Party A dials party C's number and talks privately with party C after the call is answered, and then completes the transfer or hangs up.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify whether to allow a transfer during the proceeding state of a consultation call. <ul style="list-style-type: none"> For more information, refer to SIP <SIP/> on page A-10. Specify whether a transfer is blind or not. <ul style="list-style-type: none"> For more information, refer to Call Handling Configuration <call/> on page A-58.
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Local / Centralized Conferencing

The phone can conference together the local user with the remote parties of a configurable number of independent calls by using the phone's local audio processing resources for the audio bridging. There is no dependency on network signaling for local conferences.

The phone also supports centralized conferences for which external resources are used such as a conference bridge. This relies on network signaling.

Note

Conferences are not available when the G.729 codec is enabled on the SoundStation IP 4000 conference phone.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	<p>Specify the conference hold behavior (all parties on hold or only host is on hold).</p> <ul style="list-style-type: none"> For more information, refer to Call Handling Configuration <call/> on page A-58. <p>Specify whether or not all parties hear sound effects while setting up a conference.</p> <ul style="list-style-type: none"> For more information, refer to Call Handling Configuration <call/> on page A-58. <p>Specify which type of conference to establish and the address of the centralized conference resource.</p> <ul style="list-style-type: none"> For more information, refer to Conference Setup <conference/> on page A-15.
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Manage Conferences

Note

This feature is only supported on the SoundPoint IP 550, 560, and 650 desktop phones.

This feature requires a license key for activation. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The individual parties within a conference can be managed.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	<p>Turn this feature on or off.</p> <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
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Call Forward

The phone provides a flexible call forwarding feature to forward calls to another destination. Call forwarding can be applied in the following cases:

- Automatically to all calls
- Calls from a specific caller (extension)
- When the phone is busy
- When Do Not Disturb is active
- After an extended period of alerting

The user can elect to manually forward calls while they are in the alerting state to a predefined or manually specified destination. The call forwarding feature works in conjunction with the distinctive incoming call treatment feature (refer to [Distinctive Incoming Call Treatment](#) on page 4-6). The user's ability to originate calls is unaffected by all call forwarding options. Each registration has its own forwarding properties.

Server-based call forwarding is active if the feature is enabled on both the phone and the server and the phone is registered. If server-based call forwarding is enabled on any of the phone's registrations, the other registrations are not affected.

Server-based call forwarding will behave the same as per-SIP2.1 feature with the following exceptions:

- There is no indication on the phone's user interface whether or not server-based call forwarding is active.
- If server-based call forwarding is enabled, but inactive, and the user selects the call forward soft key, the "moving arrow" icon does not appear on the user's phone (incoming calls are not forwarded).

Note

Server-based call forwarding is disabled if Shared Call Appearance or Bridged Line Appearance is enabled.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Enable or disable server-based call forwarding. <ul style="list-style-type: none"> For more information, refer to SIP <SIP/> on page A-10
	Configuration file: phone1.cfg	Enable or disable server-based call forwarding as a per-registration feature. <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89 Set all call diversion settings including a global forward-to contact and individual settings for call forward all, call forward busy, call forward no-answer, and call forward do-not-disturb. <ul style="list-style-type: none"> For more information, refer to Diversion <divert/> on page A-96.
Local	Web Server (if enabled)	Set all call diversion settings. Navigate to: <code>http://<phoneIPAddress>/reg.htm</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.
	Local Phone User Interface	The user can set the call-forward-all setting from the idle display (enable/disable and specify the forward-to contact) as well as divert callers while the call is alerting. Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Directed Call Pick-Up

Calls to another phone can be picked up by dialing the extension of the other phone. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
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Group Call Pick-Up

Calls to another phone within a pre-defined group can be picked up without dialing the extension of the other phone. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
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Call Park/Retrieve

An active call can be parked, and the parked call can be retrieved by another phone. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
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Last Call Return

The phone allows server-based last call return. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83. Specify the string sent to the server for last-call-return. For more information, refer to Call Handling Configuration <call/> on page A-58.
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Setting Up Advanced Features

This section provides information for making configuration changes for the following advanced features:

- [Configurable Feature Keys](#)
- [Multiple Line Keys per Registration](#)
- [Multiple Call Appearances](#)
- [Shared Call Appearances](#)
- [Bridged Line Appearance](#)
- [Busy Lamp Field](#)
- [Customizable Fonts and Indicators](#)

- [Instant Messaging](#)
- [Multilingual User Interface](#)
- [Downloadable Fonts](#)
- [Synthesized Call Progress Tones](#)
- [Microbrowser](#)
- [Real-Time Transport Protocol Ports](#)
- [Network Address Translation](#)
- [Corporate Directory](#)
- [Recording and Playback of Audio Calls](#)

This section also provides information for making configuration changes for the following advanced call server features:

- [Voice Mail Integration](#)
- [Multiple Registrations](#)
- [Automatic Call Distribution](#)
- [Server Redundancy](#)
- [Presence](#)
- [Microsoft Live Communications Server 2005 Integration](#)

Configurable Feature Keys

All key functions can be changed from the factory defaults. The scrolling timeout for specific keys can be configured.

Note

No feature keys on the SoundStation IP 4000 can be remapped. Since there is no Redial key on the SoundPoint IP 330/320 phone, the redial function cannot be remapped.

The rules for remapping of key functions are:

- The phone keys that have removable key caps can be mapped to the following:
 - Any function that is implemented as a removable key cap on any of the phones (Directories, Applications, Conference, Transfer, Redial, Menu, Messages, Do Not Disturb, Call Lists)
 - A speed-dial
 - Null

- The phone keys without removable key caps cannot be remapped. These include:
 - Any keys on the dial pad
 - Volume control
 - Handsfree, Mute, Headset
 - Hold
 - Navigation Cluster

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: sip.cfg	Set the key scrolling timeout, key functions, and sub-pointers for each key (usually not necessary). <ul style="list-style-type: none"> • For more information, refer to Keys <key/> on page A-69.
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For more information on the default feature key layouts, refer to [Default Feature Key Layouts](#) on page [C-10](#).

Multiple Line Keys per Registration

More than one Line Key can be allocated to a single registration (phone number or line) on SoundPoint IP phones. The number of Line Keys allocated per registration is configurable.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: phone1.cfg	Specify the number of line keys to assign per registration. <ul style="list-style-type: none"> • For more information, refer to Registration <reg/> on page A-89.
Local	Web Server (if enabled)	Specify the number of line keys to assign per registration. Navigate to <code>http://<phoneIPAddress>/reg.htm</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.
	Local Phone User Interface	Specify the number of line keys to assign per registration using the SIP Configuration menu. Either the Web Server or the boot server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the SIP Configuration menu is used, it is assumed that all registrations use the same server.

Multiple Call Appearances

The phone supports multiple concurrent calls. The hold feature can be used to pause activity on one call and switch to another call. The number of concurrent calls per line key is configurable. Each registration can have more than one line key assigned to it (refer to the previous section, [Multiple Line Keys per Registration](#)).

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the default number of calls that can be active or on hold per line key. <ul style="list-style-type: none"> For more information, refer to Call Handling Configuration <call/> on page A-58.
	Configuration file: phone1.cfg	Specify per-registration the number of calls that can be active or on hold per line key assigned to that registration. This will override the default value specified in sip.cfg . <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89.
Local	Web Server (if enabled)	Specify the default number of calls that can be active or on hold per line key and the number of calls per registration that can be active or on hold per line key assigned to that registration. Navigate to <code>http://<phoneIPAddress>/appConf.htm#ls</code> and <code>http://<phoneIPAddress>/reg.htm</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.
	Local Phone User Interface	Specify per-registration the number of calls that can be active or on hold per line key assigned to that registration using the SIP Configuration menu. Either the Web Server or the boot server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the SIP Configuration menu is used, it is assumed that all registrations use the same server.

Shared Call Appearances

Calls and lines on multiple phones can be logically related to each other. A call that is active on one phone will be presented visually to phones that share that call appearance. Mutual exclusion features emulate traditional PBX or key system privacy for shared calls. Incoming calls can be presented to multiple phones simultaneously. Users at the different locations have the ability to interrupt remote active calls.

This feature is dependent on support from a SIP server that binds the appearances together logically and looks after the necessary state notifications and performs an access control function. For more information, refer to [Shared Call Appearance Signaling](#) on page [B-10](#).

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	<p>Specify whether diversion should be disabled on shared lines.</p> <ul style="list-style-type: none"> For more information, refer to Shared Calls <shared/> on page A-60. <p>Specify line-seize subscription period.</p> <ul style="list-style-type: none"> For more information, refer to Server <server/> on page A-7. <p>Specify standard or non-standard behavior for processing line-seize subscription for mutual exclusion feature.</p> <ul style="list-style-type: none"> For more information, refer to Special Events <specialEvent/> on page A-15.
	Configuration file: phone1.cfg	<p>Specify per-registration line type (private or shared), barge-in capabilities, and line-seize subscription period if using per-registration servers. A shared line will subscribe to a server providing call state information.</p> <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89. <p>Specify per-registration whether diversion should be disabled on shared lines.</p> <ul style="list-style-type: none"> For more information, refer to Diversion <divert/> on page A-96.
Local	Web Server (if enabled)	<p>Specify line-seize subscription period.</p> <p>Navigate to <code>http://<phoneIPAddress>/appConf.htm#se</code></p> <p>Specify standard or non-standard behavior for processing line-seize subscription for mutual exclusion feature.</p> <p>Navigate to <code>http://<phoneIPAddress>/appConf.htm#ls</code></p> <p>Specify per-registration line type (private or shared) and line-seize subscription period if using per-registration servers, and whether diversion should be disabled on shared lines.</p> <p>Navigate to <code>http://<phoneIPAddress>/reg.htm</code></p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.</p>
	Local Phone User Interface	<p>Specify per-registration line type (private or shared) using the SIP Configuration menu. Either the Web Server or the boot server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the SIP Configuration menu is used, it is assumed that all registrations use the same server.</p>

Bridged Line Appearance

Calls and lines on multiple phones can be logically related to each other. A call that is active on one phone will be presented visually to phones that share that line. Mutual exclusion features emulate traditional PBX or key system privacy for shared calls. Incoming calls can be presented to multiple phones

simultaneously. This feature is dependent on support from a SIP server that binds the appearances together logically and looks after the necessary state notifications and performs an access control function. For more information, refer to [Bridged Line Appearance Signaling](#) on page B-10.

Note

In the configuration files, bridged lines are configured by “shared line” parameters.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify whether diversion should be disabled on shared lines. <ul style="list-style-type: none"> For more information, refer to Call Handling Configuration <call/> on page A-58.
	Configuration file: phone1.cfg	Specify per-registration line type (private or shared) and the shared line third party name. A shared line will subscribe to a server providing call state information. <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89. Specify per-registration whether diversion should be disabled on shared lines. <ul style="list-style-type: none"> For more information, refer to Diversion <divert/> on page A-96.
Local	Web Server (if enabled)	Specify per-registration line type (private or shared) and third party name, and whether diversion should be disabled on shared lines. Navigate to <code>http://<phoneIPAddress>/reg.htm</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.
	Local Phone User Interface	Specify per-registration line type (private or shared) and the shared line third party name using the SIP Configuration menu. Either the Web Server or the boot server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the SIP Configuration menu is used, it is assumed that all registrations use the same server.

Busy Lamp Field

Note

This feature is available only on SoundPoint IP 600 phones and SoundPoint IP 601 and 650 phones with an attached Expansion Module.

The Busy Lamp Field (BLF) feature enhances support for a phone-based attendant console. It allows monitoring the hook status and remote party information of users through the busy lamp fields and displays on an attendant console phone.



Polycom recommends that the BLF not be used in conjunction with the Microsoft Live Communications Server 2005 feature. For more information, refer to [Microsoft Live Communications Server 2005 Integration](#) on page 4-43.

Note

Use this feature with TCPpreferred transport (refer to [Server <server/>](#) on page A-7).

Configuration changes can performed centrally at the boot server:

Central (boot server)	Configuration file: phone1.cfg	Specify the list SIP URI and index of the registration which will be used to send a SUBSCRIBE to the list SIP URI specified in <code>attendant.uri</code> . <ul style="list-style-type: none"> For more information, refer to Attendant <attendant/> on page A-103.
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Customizable Fonts and Indicators

The phone's user interface can be customized by changing the fonts and graphic icons used on the display and the LED indicator patterns. Pre-existing fonts embedded in the software can be overwritten or new fonts can be downloaded. The bitmaps and bitmap animations used for graphic icons on the display can be changed and repositioned. LED flashing sequences and colors can be changed.

Configuration changes can performed centrally at the boot server:

Central (boot server)	Configuration File: sip.cfg	Specify fonts to overwrite existing ones or specify new fonts. <ul style="list-style-type: none"> For more information, refer to Fonts on page A-66. Specify which bitmaps to use. <ul style="list-style-type: none"> For more information, refer to Bitmaps <bitmap/> on page A-71. Specify how to create animations and LED indicator patterns. <ul style="list-style-type: none"> For more information, refer to Indicators <ind/> on page A-71.
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Instant Messaging

The phone supports sending and receiving instant text messages. The user is alerted to incoming messages visually and audibly. The user can view the messages immediately or when it is convenient. For sending messages, the user can either select a message from a preset list of short messages or an alphanumeric text entry mode allows the typing of custom messages using the dial pad. Message sending can be initiated by replying to an incoming

message or by initiating a new dialog. The destination for new dialog messages can be entered manually or selected from the contact directory, the preferred method.

There are no related configuration changes.

Multilingual User Interface

Note

This feature is not available on SoundPoint IP 301 phones.

The system administrator or the user can select the language. Support for major western European languages is included and additional languages can be easily added. Support for Asian languages (Chinese, Japanese, and Korean) is also included, but will display only on the SoundPoint IP 650 and SoundStation IP 4000's higher resolution displays.

For basic character support and extended character support (available on SoundPoint IP 550, 560, 600, 601, and 650 and SoundStation IP platforms), refer to [Multilingual <ml/>](#) on page A-20. (Note that within a Unicode range, some characters may not be supported due to their infrequent usage.)

Note

The multilingual feature relies on dictionary files resident on the boot server. The dictionary files are downloaded from the boot server whenever the language is changed or at boot time when a language other than the internal US English language has been configured. If the dictionary files are inaccessible, the language will revert to the internal language.

Note

Currently, the multilingual feature is only available in the application. At this time, the bootROM application is available in English only.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the boot-up language and the selection of language choices to be made available to the user. <ul style="list-style-type: none"> For more information, refer to Multilingual <ml/> on page A-20. For instructions on adding new languages, refer to To add new languages to those included with the distribution: on page A-21.
Local	Local Phone User Interface	The user can select the preferred language under the Settings menu. Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Downloadable Fonts

New fonts can be loaded onto the phone. For guidelines on downloading fonts, refer to [Fonts](#) on page [A-66](#).

Synthesized Call Progress Tones

In order to emulate the familiar and efficient audible call progress feedback generated by the PSTN and traditional PBX equipment, call progress tones are synthesized during the life cycle of a call. These call progress tones are easily configurable for compatibility with worldwide telephony standards or local preferences.

Configuration changes can be performed centrally at the boot server:

<p>Central (boot server)</p>	<p>Configuration file: sip.cfg</p>	<p>Specify the basic tone frequencies, levels, and basic repetitive cadences.</p> <ul style="list-style-type: none"> For more information, refer to Chord-Sets <chord/> on page A-26. <p>Specify downloaded sampled audio files for advanced call progress tones.</p> <ul style="list-style-type: none"> For more information, refer to Sampled Audio for Sound Effects <saf/> on page A-27. <p>Specify patterns.</p> <ul style="list-style-type: none"> For more information, refer to Patterns <pat/> on page A-29 and Call Progress Patterns on page A-30.
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Microbrowser

The SoundPoint IP 430, 501, 550, 560, 600, 601, and 650 phones and the SoundStation IP 4000 phone supports an XHTML Microbrowser. This can be launched by pressing the **Applications** key, or if there isn't one on the phone, it can be accessed through the **Menu** key by selecting *Features*, and then *Applications*.

Note

As of SIP 2.2.0, the **Services** key and menu entry are renamed **Applications**, however the functionality remains the same.

Two instances of the Microbrowser may run concurrently:

- An instance with standard interactive user interface
- An instance that does not support user input, but appears in a window on the idle display

For more information, refer to the *Web Application Developers's Guide*.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the Application browser home page, a proxy to use, and size limits. • For more information, refer to Microbrowser <mb/> on page A-86.
Local	Web Server (if enabled)	Specify the Applications browser home page and proxy to use. Navigate to <code>http://<phoneIPAddress>/coreConf.htm#mb</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Real-Time Transport Protocol Ports

The phone is compatible with RFC 1889 - RTP: A Transport Protocol for Real-Time Applications - and the updated RFCs 3550 and 3551. Consistent with RFC 1889, the phone treats all RTP streams as bi-directional from a control perspective and expects that both RTP end points will negotiate the respective destination IP addresses and ports. This allows real-time transport control protocol (RTCP) to operate correctly even with RTP media flowing in only a single direction, or not at all. It also allows greater security: packets from unauthorized sources can be rejected.

The phone can filter incoming RTP packets arriving on a particular port by IP address. Packets arriving from a non-negotiated IP address can be discarded.

The phone can also enforce symmetric port operation for RTP packets: packets arriving with the source port set to other than the negotiated remote sink port can be rejected.

The phone can also jam the destination transport port to a specified value regardless of the negotiated port. This can be useful for punching through firewalls. When this is enabled, all RTP traffic will be sent to the specified port and will be expected to arrive on that port as well. Incoming packets are sorted by the source IP address and port, allowing multiple RTP streams to be multiplexed.

The RTP port range used by the phone can be specified. Since conferencing and multiple RTP streams are supported, several ports can be used concurrently. Consistent with RFC 1889, the next higher odd port is used to send and receive RTCP.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify whether to filter incoming RTP packets by IP address, whether to require symmetric port usage, whether to jam the destination port and specify the local RTP port range start. <ul style="list-style-type: none"> For more information, refer to RTP <rtp/> on page A-51.
Local	Web Server (if enabled)	Specify whether to filter incoming RTP packets by IP address, whether to require symmetric port usage, whether to jam the destination port and specify the local RTP port range start. Navigate to: <code>http://<phoneIPAddress>/netConf.htm#rt</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Network Address Translation

The phone can work with certain types of network address translation (NAT). The phone's signaling and RTP traffic use symmetric ports (the source port in transmitted packets is the same as the associated listening port used to receive packets) and the external IP address and ports used by the NAT on the phone's behalf can be configured on a per-phone basis.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the external NAT IP address and the ports to be used for signaling and RTP traffic. <ul style="list-style-type: none"> For more information, refer to Network Address Translation <nat/> on page A-102.
Local	Web Server (if enabled)	Specify the external NAT IP address and the ports to be used for signaling and the RTP traffic. Navigate to: <code>http://<phoneIPAddress>/netConf.htm#na</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Corporate Directory

Note

This feature requires a license key for activation. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The SoundPoint IP phones can be configured to interface with a corporate directory server that supports the Lightweight Directory Access Protocol (LDAP) version 3. (Microsoft’s Active Directory is included.) Both corporate directories that support server-side sorting and those that do not are supported. In the latter case, the sorting is performed on the phone.



Polycom recommends using corporate directories that have server-side sorting. Polycom recommends that you consult your LDAP Administrator when making any configuration changes for this feature.

The corporate directory can be browsed or searched. Entries retrieved from the LDAP server can be saved to the local contact directory on the phone. Phone calls can be placed based on the phone number contained in the LDAP entry.

The corporate directory interface shall be read only, so that editing or deleting existing directory entries as well as adding new directory entries from the phone shall not be possible.

All attributes are considered to be Unicode text. Validity checking will be performed when a call is placed or the entry is saved to the local contact directory.

The corporate directory LDAP server status can be reviewed through the Status menu (*Status > CD Server Status*).

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify the location of the corporate directory’s LDAP server, the LDAP attributes, how often to refresh the local cache from the LDAP server, and other miscellaneous parameters. <ul style="list-style-type: none"> For more information, refer to Corporate Directory <corp/> on page A-63.
Local	Local Phone User Interface	Enable or disable persistent viewing through the Settings menu (<i>Settings > Basic > Preferences > Corporate Directory > View Persistency</i>). Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

This section contains the following information:

- [Corporate Directory LDAP Attributes](#)
- [Browsing the Corporate Directory](#)
- [Configuration File Example](#)

Corporate Directory LDAP Attributes

The entry attributes in the corporate directory are mapped through **sip.cfg** configuration file attributes to the LDAP attributes `first_name`, `last_name`, `phone_number`, and others so the SIP application knows how to use them for searching, dialing, or saving to the local contact directory. Multiple attributes of the same type are allowed.

The configuration order dictates how the attributes are displayed and sorted. The first attribute is the primary sort index and the second attribute is the secondary sort index. The other attributes are not used in sorting.

To limit the amount of data displayed in the corporate directory, filtering of the entries can be configured for all attribute types. Filtering can be configured to be retained if the phone reboots.

For more information on LDAP attributes, refer to *RFC 4510 - Lightweight Directory Access Protocol (LDAP): Technical Specification Road Map*.

Browsing the Corporate Directory

The SoundPoint IP or SoundStation IP phone will establish a session with the corporate directory and download enough entries to fill its cache:

- when the corporate directory is first accessed
- when the phone boots up if the background synchronization parameter is enabled

The requested entries are based on the configured attributes (see previous section).

If the background synchronization parameter is enabled, a timer is initiated to permit a periodic download from the corporate directory.

Entries are sorted according to the order in which the first two attributes are configured (for example, last name, then first name).

The browse position within the corporate directory as well as the attribute filters are maintained for subsequent corporate directory access can be saved (if so configured).

Configuration File Example

The following excerpt from the **sip.cfg** configuration file shows an example where downloaded entries are limited to any where the phone number is in the 604 area code:

```
dir.corp.address=""
dir.corp.port=""
dir.corp.transport="TCP"
dir.corp.baseDN="cn=Users,dc=yourcompany,dc=local
dir.corp.user="ldapadmin"
dir.corp.password="12345678"
```

```

dir.corp.filterPrefix="(objectclass=person)"
dir.corp.attribute.1.name="sn"
dir.corp.attribute.1.label="Last Name"
dir.corp.attribute.1.type="last_name"
dir.corp.attribute.1.filter=""
dir.corp.attribute.1.sticky=""
dir.corp.attribute.2.name="givenName"
dir.corp.attribute.2.label="First Name"
dir.corp.attribute.2.type="first_name"
dir.corp.attribute.2.filter=""
dir.corp.attribute.2.sticky=""
dir.corp.attribute.3.name="telephoneNumber"
dir.corp.attribute.3.label="Phone Number"
dir.corp.attribute.3.type="phone_number"
dir.corp.attribute.3.filter="604"
dir.corp.attribute.3.sticky="1"
dir.corp.backGroundSync="1"
dir.corp.backGroundSync.period="60"
dir.corp.viewPersistence="1"

```

Recording and Playback of Audio Calls

Note

This feature requires a license key for activation. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

SoundPoint IP phones that have a USB port can be configured to allow recording of audio calls on a supported USB device. Currently only the SoundPoint IP 650 has a USB port.

The filenames of the recorded **.wav** files will include a date/time stamp (for example, **20Apr2007_190012.wav** was created on April 20, 2007 at 19:00:12). An indication of the recording time remaining—the space available of the attached USB storage media—appears on the graphic display. The user can browse through all recorded files through the menu shown on the graphic display.

Playback of recorded files can occur on the phone as well as on other devices, such as a Windows® or Apple® based computer using an application like Windows Media Player® or iTunes®.

The user controls which calls are recorded and played back.

For more information, refer to “Technical Bulletin 38084: SoundPoint IP 650 Supported USB Devices for Recording” at <http://www.polycom.com/support/voip/>.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
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Voice Mail Integration

The phone is compatible with voice mail servers. The subscribe contact and callback mode can be configured per user/registration on the phone. The phone can be configured with a SIP URL to be called automatically by the phone when the user elects to retrieve messages. Voice mail access can be configured to be through a single key press (for example, the Messages key on the SoundPoint IP 430, 500, 501, 550, 560, 600, 601, and 650). A message-waiting signal from a voice mail server triggers the message-waiting indicator to flash and the call waiting audio tone is played through the active audio path.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	For one-touch voice mail access, enable the “one-touch voice mail” user preference. <ul style="list-style-type: none"> For more information, refer to User Preferences <up/> on page A-23.
	Configuration file: phone1.cfg	For one-touch voice mail access, bypass instant messages to remove the step of selecting between instant messages and voice mail after pressing the Messages key on the SoundPoint IP 430, 500, 501, 550, 560, 600, 601, and 650 (instant messages are still accessible from the Main Menu). On a per-registration basis, specify a subscribe contact for solicited NOTIFY applications, a callback mode (self call-back or another contact), and the contact to call when the user accesses voice mail. <ul style="list-style-type: none"> For more information, refer to Messaging <msg/> on page A-101.

Local	Web Server (if enabled)	<p>For one-touch voice mail access, enable the “one-touch voice mail” user preference and bypass instant messages to remove the step of selecting between instant messages and voice mail after pressing the Messages key on the SoundPoint IP 430, 500, 501, 550, 560, 600, 601, and 650 (instant messages are still accessible from the Main Menu).</p> <p>Navigate to <code>http://<phoneIPAddress>/coreConf.htm#us</code></p> <p>On a per-registration basis, specify a subscribe contact for solicited NOTIFY applications, a callback mode (self call-back or another contact) to call when the user accesses voice mail.</p> <p>Navigate to <code>http://<phoneIPAddress>/reg.htm</code></p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <code><Ethernet address>-phone.cfg</code> is removed from the boot server.</p>
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Multiple Registrations

The SoundPoint IP 301, 320, 330, and 430 support a maximum of two registrations, the SoundPoint IP 501 supports three, the SoundPoint IP 550 and 560 supports four, and the SoundPoint IP 600, 601, and 650 support 6. Up to three SoundPoint IP Expansion Modules can be added to a single host SoundPoint IP 601 and 650 phone increasing the total number of buttons to 12 registrations on the IP 601 and 34 registrations on the IP 650. The SoundStation IP 4000 supports a single registration.

Each registration can be mapped to one or more line keys (a line key can be used for only one registration). The user can select which registration to use for outgoing calls or which to use when initiating new instant message dialogs.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	<p>Specify the local SIP signaling port and an array of SIP servers to register to. For each server specify the registration period and the signaling failure behavior.</p> <ul style="list-style-type: none"> For more information, refer to Local <local/> on page A-6 and Server <server/> on page A-7.
	Configuration file: phone1.cfg	<p>For up to maximum number of registrations, specify a display name, a SIP address, an optional display label, an authentication user ID and password, the number of line keys to use, and an optional array of registration servers. The authentication user ID and password are optional and for security reasons can be omitted from the configuration files. The local flash parameters will be used instead. The optional array of servers and their associated parameters will override the servers specified in sip.cfg if non-Null.</p> <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89.

Local	Web Server (if enabled)	<p>Specify the local SIP signaling port and an array of SIP servers to register to.</p> <p>Navigate to <code>http://<phoneIPAddress>/appConf.htm#se</code></p> <p>For up to six registrations (depending on the phone model, in this case the maximum is six even for the IP 601 and 650), specify a display name, a SIP address, an optional display label, an authentication user ID and password, the number of line keys to use, and an optional array of registration servers. The authentication user ID and password are optional and for security reasons can be omitted from the configuration files. The local flash parameters will be used instead. The optional array of servers will override the servers specified in sip.cfg in non-Null. This will also override the servers on the appConf.htm web page.</p> <p>Navigate to <code>http://<phoneIPAddress>/reg.htm</code></p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.</p>
Local (continued)	Local Phone User Interface	<p>Use the SIP Configuration menu to specify the local SIP signaling port, a default SIP server to register to and registration information for up to twelve registrations (depending on the phone model). The SIP Configuration menu contains a sub-set of all the parameters available in the configuration files.</p> <p>Either the Web Server or the boot server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the SIP Configuration menu is used, it is assumed that all registrations use the same server.</p> <p>Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.</p> <p>For more information, refer to Local <local/> on page A-6, Server <server/> on page A-7, and Registration <reg/> on page A-89.</p>

Automatic Call Distribution

The phone allows automatic call distribution (ACD) login and logout. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	<p>Turn this feature on or off.</p> <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
	Configuration file: phone1.cfg	<p>Enable this feature per registration.</p> <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89.

The phone also supports ACD agent available and unavailable. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
	Configuration file: phone1.cfg	Enable this feature per registration. <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89.

Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service for events where the call server needs to be taken offline for maintenance, the server fails, or the connection from the phone to the server fails.

Two types of redundancy are possible:

- **Fail-over:** In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using DNS mechanisms or “IP Address Moving” from the primary to the back-up server.
- **Fallback:** In this mode, a second less featured call server (router or gateway device) with SIP capability takes over call control to provide basic calling capability, but without some of the richer features offered by the primary call server (for example, shared lines, presence, and Message Waiting Indicator). Polycom phones support configuration of multiple servers per SIP registration for this purpose.

In some cases, a combination of the two may be deployed.

Note

Your SIP server provider should be consulted for recommended methods of configuring phones and servers for fail-over configuration.

Warning

Prior to SIP 2.1, the `reg.x.server.y` parameters (refer to [Registration <reg/>](#) on page [A-89](#)) could be used for fail-over configuration. The older behavior is no longer supported. Customers that are using the `reg.x.server.y` configuration parameters where $y \geq 2$ should take care to ensure that their current deployments are not adversely affected. For example the phone will only support advanced SIP features such as shared lines, missed calls, presence with the primary server ($y=1$).

For more information, refer to “Technical Bulletin 5844: SIP Server Fallback Enhancements on SoundPoint IP Phones” at <http://www.polycom.com/support/voice/>.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify global primary and fallback server configuration parameters. <ul style="list-style-type: none"> For more information, refer to Protocol <volpProt/> on page A-6.
	Configuration file: phone1.cfg	Specify per registration primary and fallback server configuration parameters values that override those in sip.cfg . <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89.

DNS SIP Server Name Resolution

If a DNS name is given for a proxy/registrar address, the IP address(es) associated with that name will be discovered as specified in RFC 3263. If a port is given, the only lookup will be an A record. If no port is given, NAPTR and SRV records will be tried, before falling back on A records if NAPTR and SRV records return no results. If no port is given, and none is found through DNS, 5060 will be used.

Refer to <http://www.ietf.org/rfc/rfc3263.txt> for an example.

Note

Failure to resolve a DNS name is treated as signalling failure that will cause a failover.

Behavior When the Primary Server Connection Fails

For Outgoing Calls (INVITE Fallback)

When the user initiates a call, the phone will go through the following steps to connect the call:

1. Try to make the call using the working server.
2. If the working server does not respond correctly to the INVITE, then try and make a call using the next server in the list (even if there is no current registration with these servers). This could be the case if the Internet connection has gone down, but the registration to the working server has not yet expired.
3. If the second server is also unavailable, the phone will try all possible servers (even those not currently registered) until it either succeeds in making a call or exhausts the list at which point the call will fail.

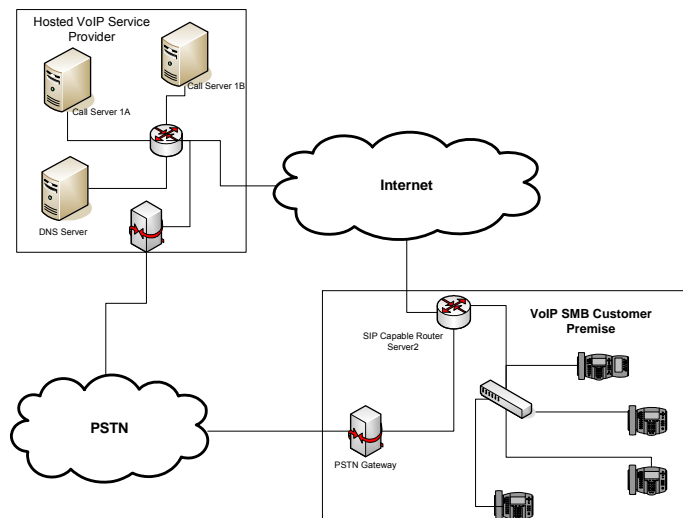
At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below:

- If TCP is used, then the signaling fails if the connection fails or the Send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list and this is the last server, then the signaling fails after the complete UDP

timeout defined in RFC 3261. If it is not the last server in the list, the maximum number of retries using the configurable retry timeout is used. For more information, refer to [Server <server/>](#) on page A-7 and [Registration <reg/>](#) on page A-89.

Warning

If DNS is used to resolve the address for Servers, the DNS server is unavailable, and the TTL for the DNS records has expired, the phone will attempt to contact the DNS server to resolve the address of all servers in its list *before* initiating a call. These attempts will timeout, but the timeout mechanism can cause long delays (for example, two minutes) before the phone call proceeds “using the working server”. To mitigate this issue, long TTLs should be used. It is strongly recommended that an on-site DNS server is deployed as part of the redundancy solution.



Phone Configuration

The phones at the customer site are configured as follows:

- Server 1 (the primary server) will be configured with the address of the service provider call server. The IP address of the server(s) to be used will be provided by the DNS server. For example:

```
reg.1.server.1.address="voipserver.serviceprovider.com"
```

- Server 2 (the fallback server) will be configured to the address of the router/gateway that provides the fallback telephony support and is on-site. For example:

```
reg.1.server.2.address=172.23.0.1
```

Note

It is possible to configure the phone for more than two servers per registration, but you need to exercise caution when doing this to ensure that the phone and network load generated by registration refresh of multiple registrations do not become excessive. This would be of particular concern if a phone had multiple registrations with multiple servers per registration and it is expected that some of these servers will be unavailable.

Phone Operation for Registration

After the phone has booted up, it will register to all the servers that are configured.

Server 1 is the primary server and supports greater SIP functionality than any of servers. For example, SUBSCRIBE/NOTIFY services (used for features such as shared lines, presence, and BLF) will only be established with Server 1.

Upon registration timer expiry of each server registration, the phone will attempt to re-register. If this is unsuccessful, normal SIP re-registration behavior (typically at intervals of 30 to 60 seconds) will proceed and continue until the registration is successful (for example, when the Internet link is once again operational). While the primary server registration is unavailable, the next highest priority server in the list will serve as the working server. As soon as the primary server registration succeeds, it will return to being the working server.

Note

If `reg.x.server.y.register` is set to 0, then phone will not register to that server. However, the INVITE will fail over to that server if all higher priority servers are down.

Recommended Practices for Fallback Deployments

In situations where server redundancy for fall-back purpose is used, the following measures should be taken to optimize the effectiveness of the solution:

1. Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.
2. Do not use OutBoundProxy configurations on the phone if the OutBoundProxy could be unreachable when the fallback occurs. SoundPoint IP phones can only be configured with one OutBoundProxy per registration and all traffic for that registration will be routed through this proxy for all servers attached to that registration. If Server 2 is not accessible through the configured proxy, call signaling with Server 2 will fail.
3. Avoid using too many servers as part of the redundancy configuration as each registration will generate more traffic.
4. Educate users as to the features that will not be available when in "fallback" operating mode.

Presence

The Presence feature allows the phone to monitor the status of other users/devices and allows other users to monitor it. The status of monitored users is displayed visually and is updated in real time in the Buddies display screen or, for speed dial entries, on the phone's idle display. Users can block others from monitoring their phones and are notified when a change in monitored status occurs. Phone status changes are broadcast automatically to monitoring phones when the user engages in calls or invokes do-not-disturb. The user can also manually specify a state to convey, overriding, and perhaps masking, the automatic behavior.

Note

Notification when a change in monitored status occurs will be available in a subsequent release.

The presence feature works differently when Microsoft Live Communications Server 2005 is used as the call server. For more information, refer to the following section, [Microsoft Live Communications Server 2005 Integration](#).

Configuration changes can be performed centrally at the boot server:

Central (boot server)	XML file: <Ethernet address>-directory.xml	The <code><bw></code> (buddy watching) and <code><bb></code> (buddy blocking) elements in the <Ethernet address>-directory.xml file dictate the Presence aspects of directory entries. <ul style="list-style-type: none"> For more information, refer to Local Contact Directory on page 4-9.
Local	Local Phone User Interface	The user can edit the directory contents. The <i>Watch Buddy</i> and <i>Block Buddy</i> fields control the buddy behavior of contacts. Changes will be stored in the phone's flash file system and backed up to the boot server copy of <Ethernet address>-directory.xml if this is configured. When the phone boots, the boot server copy of the directory, if present, will overwrite the local copy.

Microsoft Live Communications Server 2005 Integration

SoundPoint IP phones can be used with Microsoft Live Communications Server 2005 and Microsoft Office Communicator to help improve business efficiencies and increase productivity and to share ideas and information immediately with business contacts.

For instructions on changing the configuration files, refer to [Configuration File Examples](#) on page 4-44.

Note

Any contacts added through the SoundPoint IP phone's buddy list will appear in as a contact in Microsoft Office Communicator and Windows Messenger.



Polycom recommends that the BLF not be used in conjunction with the Microsoft Live Communications Server 2005 feature. For more information, refer to [Busy Lamp Field](#) on page 4-27.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify that support for Microsoft Live Communications Server 2005 is enabled. <ul style="list-style-type: none"> For more information, refer to SIP <SIP/> on page A-10. Specify the line/registration number used to send SUBSCRIBE for presence. <ul style="list-style-type: none"> For more information, refer to Presence <pres/> on page A-65. Turn the presence and messaging features on or off. <ul style="list-style-type: none"> For more information, refer to Feature <feature/> on page A-83.
	Configuration file: phone1.cfg	Specify the number of line keys to assign per registration. <ul style="list-style-type: none"> For more information, refer to Registration <reg/> on page A-89. Specify the line/registration number which has roaming buddies support enabled. <ul style="list-style-type: none"> For more information, refer to Roaming Buddies <roaming_buddies/> on page A-104. Specify the line/registration number which has roaming privacy support enabled. <ul style="list-style-type: none"> For more information, refer to Roaming Privacy <roaming_privacy/> on page A-104.

Configuration File Examples

SoundPoint IP phones can be deployed in two basic methods. In the first method, Microsoft Live Communications Server 2005 serves as the call server and the phones have a single registration. In the second method, the phone has a primary registration to call server—that is not Microsoft Live Communications Server (LCS)—and a secondary registration to LCS for presence purposes.

To set up a single registration with Microsoft Live Communications Server 2005 as the call server:

1. Modify the **sip.cfg** configuration file as follows:
 - a Open **sip.cfg** in an XML editor.
 - b Locate the feature parameter.
 - c For the `feature.1.name = presence` attribute, set `feature.1.enabled` to 1.

- d** For the `feature.2.name = messaging` attribute, set `feature.2.enabled` to 1.
- e** Locate the `voIpProt` parameter.
Set the `voIpProt.server.x.transport` attribute to `TCPpreferred` or `TLS`.
Your selection depends on the LCS configuration.

Note

The TLS protocol is not supported on SoundPoint IP 300 and 500 phones.

- f** Set the `voIpProt.server.x.address` to the LCS address.
For example, `voIpProt.server.1.address = "lcs2005.local"`
 - g** Set the `voIpProt.SIP.lcs` attribute to 1.
 - h** (Optional) If SIP forking is desired, set `voIpProt.SIP.ms-forking` attribute to 1.
Refer to [SIP <SIP/>](#) on page [A-10](#).
 - i** Save the modified `sip.cfg` configuration file.
- 2.** Modify the `phone1.cfg` configuration file as follows:
- a** Open `phone1.cfg` in an XML editor.
 - b** Locate the registration parameter.
 - c** Set the `reg.1.address` to the LCS address.
For example, `reg.1.address = "7778"`
 - d** Set the `reg.1.server.y.address` to the LCS server name.
 - e** (Optional) Set the `reg.1.server.y.transport` attribute to `TCPpreferred` or `TLS`.
Your selection depends on the LCS configuration.
 - f** Set `reg.1.auth.userId` to the phone's LCS username.
For example, `reg.1.auth.userId = "jbloggs"`
 - g** Set `reg.1.auth.password` to the LCS password.
For example, `reg.1.auth.password = "Password2"`
 - h** Locate the `roaming_buddies` attribute.
 - i** Set the `roaming_buddies.reg` element to 1.
Refer to [Roaming Buddies <roaming_buddies/>](#) on page [A-104](#).
 - j** Locate the `roaming_privacy` attribute.
 - k** Set the `roaming_privacy.reg` element to 1.
Refer to [Roaming Privacy <roaming_privacy/>](#) on page [A-104](#).

- l** Save the modified **phone1.cfg** configuration file.

To set up a dual registration with Microsoft Live Communications Server 2005 as the presence server:

- 1.** (Optional) Modify the **sip.cfg** configuration file as follows:
 - a** Open **sip.cfg** in an XML editor.
 - b** Locate the feature parameter.
 - c** For the `feature.1.name = presence` attribute, set `feature.1.enabled` to 1.
 - d** For the `feature.2.name = messaging` attribute, set `feature.2.enabled` to 1.
 - e** Locate the `voIpProt` parameter.
 - f** If SIP forking is desired, set `voIpProt.SIP.ms-forking` attribute to 1.
Refer to [SIP <SIP/>](#) on page A-10.
 - g** Save the modified **sip.cfg** configuration file.
- 2.** Modify the **phone1.cfg** configuration file as follows:
 - a** Open **phone1.cfg** in an XML editor.
 - b** Locate the registration parameter.
 - c** Select a registration to be used for the Microsoft Live Communications Server 2005.
Typically, this would be 2.
 - d** Set the `reg.x.address` to the LCS address.
For example, `reg.2.address = "7778"`
 - e** Set the `reg.x.server.y.address` to the LCS server name.
 - f** (Optional) Set the `reg.2.server.y.transport` attribute to TCPpreferred or TLS.
Your selection depends on the LCS configuration.
 - g** Set `reg.x.auth.userId` to the phone's LCS username.
For example, `reg.2.auth.userId = "jbloggs"`
 - h** Set `reg.x.auth.password` to the LCS password.
For example, `reg.2.auth.password = "Password2"`
 - i** Locate the `roaming_buddies` attribute.
 - j** Set the `roaming_buddies.reg` element to the number corresponding to the LCS registration.
For example, `roaming_buddies.reg = 2`
Refer to [Roaming Buddies <roaming_buddies/>](#) on page A-104.

- k** Locate the `roaming_privacy` attribute.
- l** Set the `roaming_privacy.reg` element to the number corresponding to the LCS registration.
For example, `roaming_privacy.reg = 2`
Refer to [Roaming Privacy <roaming_privacy/>](#) on page [A-104](#).
- m** Save the modified `phone1.cfg` configuration file.

Setting Up Audio Features

Proprietary state-of-the-art digital signal processing (DSP) technology is used to provide an excellent audio experience.

This section provides information for making configuration changes for the following audio-related features:

- [Low-Delay Audio Packet Transmission](#)
- [Jitter Buffer and Packet Error Concealment](#)
- [Voice Activity Detection](#)
- [DTMF Tone Generation](#)
- [DTMF Event RTP Payload](#)
- [Acoustic Echo Cancellation](#)
- [Audio Codecs](#)
- [Background Noise Suppression](#)
- [Comfort Noise Fill](#)
- [Automatic Gain Control](#)
- [IP Type-of-Service](#)
- [IEEE 802.1p/Q](#)
- [Voice Quality Monitoring](#)

Low-Delay Audio Packet Transmission

The phone is designed to minimize latency for audio packet transmission. There are no related configuration changes.

Jitter Buffer and Packet Error Concealment

The phone employs a high-performance jitter buffer and packet error concealment system designed to mitigate packet inter-arrival jitter and out-of-order or lost (lost or excessively delayed by the network) packets. The jitter buffer is adaptive and configurable for different network environments. When packets are lost, a concealment algorithm minimizes the resulting negative audio consequences.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Set the jitter buffer tuning parameters including minimum and maximum size and shrink aggression. <ul style="list-style-type: none"> For more information, refer to Codec Profiles <audioProfile/> on page A-36.
Local	Web Server (if enabled)	Set the jitter buffer tuning parameters including minimum and maximum size and shrink aggression. Navigate to <code>http://<phoneIPAddress>/coreConf.htm#au</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Voice Activity Detection

The purpose of voice activity detection (VAD) is to conserve network bandwidth by detecting periods of relative “silence” in the transmit data path and replacing that silence efficiently with special packets that indicate silence is occurring. For those compression algorithms without an inherent VAD function, such as G.711, the phone is compatible with the comprehensive codec-independent comfort noise transmission algorithm specified in RFC 3389. This algorithm is derived from G.711 Appendix II, which defines a comfort noise (CN) payload format (or bit-stream) for G.711 use in packet-based, multimedia communication systems. The phone generates CN packets (also known as Silence Insertion Descriptor (SID) frames) and also decodes CN packets, efficiently regenerating a facsimile of the background noise at the remote end.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Enable or disable VAD and set the detection threshold. <ul style="list-style-type: none"> For more information, refer to Voice Activity Detection <vad/> on page A-46.
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DTMF Tone Generation

The phone generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad. These tones are transmitted in the real-time transport protocol (RTP) streams of connected calls. The phone can encode the DTMF tones using the active voice codec or using RFC 2833 compatible encoding. The coding format decision is based on the capabilities of the remote end point.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Set the DTMF tone levels, autodialing on and off times, and other parameters. <ul style="list-style-type: none"> For more information, refer to Dual Tone Multi-Frequency <DTMF/> on page A-25.
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DTMF Event RTP Payload

The phone is compatible with RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*. RFC 2833 describes a standard RTP-compatible technique for conveying DTMF dialing and other telephony events over an RTP media stream. The phone generates RFC 2833 (DTMF only) events but does not regenerate, nor otherwise use, DTMF events received from the remote end of the call.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Enable or disable RFC 2833 support in SDP offers and specify the payload value to use in SDP offers. <ul style="list-style-type: none"> For more information, refer to Dual Tone Multi-Frequency <DTMF/> on page A-25.
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Acoustic Echo Cancellation

The phone employs advanced acoustic echo cancellation (AEC) for hands-free operation. Both linear and non-linear techniques are employed to aggressively reduce echo yet provide for natural full-duplex communication patterns.

When using the handset on any SoundPoint IP phones, AEC is not normally required. In certain situations, where echo is experienced by the far-end party, when the user is on the handset, AEC may be enabled to reduce/avoid this echo. To achieve this, make the following changes in the **sip.cfg** configuration file (default settings for these parameters are disabled):

```
voice.aec.hs.enable = 1
voice.aes.hs.enable = 1
voice.ns.hs.enable = 1
voice.ns.hs.signalAttn = -6
voice.ns.hs.silenceAttn = -9
```

For more information, refer to [Acoustic Echo Cancellation <aec/>](#) on page A-34, [Acoustic Echo Suppression <aes/>](#) on page A-41, and [Background Noise Suppression <ns/>](#) on page A-42.

For the SoundPoint IP 501 and 601, utilizing acoustic echo cancellation will introduce a small delay increase into the audio path which might cause a lower voice quality.

Note

AEC on the SoundPoint IP 301 handset is not supported.

Audio Codecs

The following table summarizes the phone's audio codec support:

Algorithm	MIME Type	Ref.	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
G.711 μ -law	PMCU	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.711a-law	PCMA	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.722	G722/8000	RFC 1890	64 Kbps	16 Ksps	10ms - 80ms	7 KHz
G.722.1	G722/16000	RFC 3047	16 Kbps, 24 Kbps, 32 Kbps	16 Ksps	20ms - 80ms	7 KHz
G.729AB	G729	RFC 1890	8 Kbps	8 Ksps	10ms - 80ms	3.5KHz
SID	CN	RFC 3389	N/A	N/A	N/A	N/A
RFC 2833	phone-event	RFC 2833	N/A	N/A	N/A	N/A

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. <ul style="list-style-type: none"> For more information, refer to Codec Preferences <codecPref/> on page A-35 and Codec Profiles <audioProfile/> on page A-36.
Local	Web Server (if enabled)	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. Navigate to <code>http://<phoneIPAddress>/coreConf.htm#au</code> Changes are saved to local flash and backed up to <Ethernet address>-phone.cfg on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the <Ethernet address>-phone.cfg is removed from the boot server.

Background Noise Suppression

Background noise suppression (BNS) is designed primarily for hands-free operation and reduces background noise to enhance communication in noisy environments.

There are no related configuration changes.

Comfort Noise Fill

Comfort noise fill is designed to help provide a consistent noise level to the remote user of a hands-free call. Fluctuations in perceived background noise levels are an undesirable side effect of the non-linear component of most AEC systems. This feature uses noise synthesis techniques to smooth out the noise level in the direction toward the remote user, providing a more natural call experience.

There are no related configuration changes.

Automatic Gain Control

Automatic Gain Control (AGC) is applicable to hands-free operation and is used to boost the transmit gain of the local talker in certain circumstances. This increases the effective user-phone radius and helps with the intelligibility of soft-talkers.

There are no related configuration changes.

IP Type-of-Service

The “type of service” field in an IP packet header consists of four type-of-service (TOS) bits and a 3-bit precedence field. Each TOS bit can be set to either 0 or 1. The precedence field can be set to a value from 0 through 7. The type of service can be configured specifically for RTP packets and call control packets, such as SIP signaling packets.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify protocol-specific IP TOS settings. <ul style="list-style-type: none"> For more information, refer to IP TOS <IP/> on page A-51.
Local	Web Server (if enabled)	Specify IP TOS settings. Navigate to: <code>http://<phoneIPAddress>/netConf.htm#qo</code>

IEEE 802.1p/Q

The phone will tag all Ethernet packets it transmits with an 802.1Q VLAN header for one of the following reasons:

- When it has a valid VLAN ID set in its network configuration
- When it is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch
- When a VLAN ID is obtained from DHCP (refer to [DHCP Menu](#) on page 3-7)

The 802.1p/Q user_priority field can be set to a value from 0 to 7. The user_priority can be configured specifically for RTP packets and call control packets, such as SIP signaling packets, with default settings configurable for all other packets.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify default and protocol-specific 802.1p/Q settings. <ul style="list-style-type: none"> • For more information, refer to Ethernet IEEE 802.1p/Q <ethernet/> on page A-50.
Local	Web Server (if enabled)	Specify 802.1p/Q settings. Navigate to http://<phoneIPAddress>/netConf.htm#qo
	Local Phone User Interface	Specify whether CDP is to be used or manually set the VLAN ID or configure DHCP VLAN Discovery. Phase 1: bootRom - Navigate to: SETUP menu during auto-boot countdown. Phase 2: Application - Navigate to: Menu>Settings>Advanced>Admin Settings>Network Configuration <ul style="list-style-type: none"> • For more information, refer to Setting Up the Network on page 3-2.

Voice Quality Monitoring

Note

This feature requires a license key for activation. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The SoundPoint IP phones can be configured to generate various quality metrics for listening and conversational quality. These metrics can be sent between the phones in RTCP XR packets. The metrics can also be downloaded in SIP messages to a central voice quality report collector. The collection of these metrics is supported on the SoundPoint IP 330/320, 430, 501, 550, 560, 600, 601, and 650 and the SoundStation IP 4000.

The RTCP XR packets are compliant with *RFC 3611 - RTP Control Extended Reports (RTCP XR)*. The packets are sent to a report collector as specified in draft RFC *draft-ietf_sipping_rtcp-summary-02*.

Three types of quality reports can be enabled:

- **Alert**—Generated when the call quality degrades below a configurable threshold.
- **Periodic**—Generated during a call at a configurable period.
- **Session**—Generated at the end of a call.

A wide range of performance metrics are generated. Some are based on current values, such as jitter buffer nominal delay and round trip delay, while others cover the time period from the beginning of the call until the report is sent, such as network packet loss. Some metrics are computed using other metrics as input, such as listening Mean Opinion Score (MOS), conversational MOS, listening R-factor, and conversational R-factor.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify the location of the central report collector, how often the reports are generated, and the warning and critical threshold values that will cause generation of alert reports. <ul style="list-style-type: none"> • For more information, refer to Quality Monitoring <quality monitoring/> on page A-47.
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Setting Up Security Features

This section provides information for making configuration changes for the following security-related features:

- [Local User and Administrator Privilege Levels](#)
- [Custom Certificates](#)
- [Incoming Signaling Validation](#)
- [Configuration File Encryption](#)

Local User and Administrator Privilege Levels

Several local settings menus are protected with two privilege levels, user and administrator, each with its own password. The phone will prompt for either the user or administrator password before granting access to the various menu options. When the user password is requested, the administrator password will also work. The web server is protected by the administrator password (refer to [Configuring SoundPoint IP / SoundStation IP Phones Locally](#) on page [4-55](#)).

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify the minimum lengths for the user and administrator passwords. <ul style="list-style-type: none"> For more information, refer to Password Lengths <pwd/><length/> on page A-80.
Local	Web Server (if enabled)	None.
	Local Phone User Interface	The user and administrator passwords can be changed under the Settings menu or through configuration parameters (refer to Flash Parameter Configuration on page A-105). Passwords can consist of ASCII characters 32-127 (0x20-0x7F) only. Changes are saved to local flash but are not backed up to <Ethernet address>-phone.cfg on the boot server for security reasons.

Custom Certificates

The phone trusts certificates issued by widely recognized certificate authorities when trying to establish a connection to a boot server for application provisioning. Refer to [Trusted Certificate Authority List](#) on page C-1.

In addition, custom certificates can be added to the phone. This is done by using the SSL Security menu on the phone to provide the URL of the custom certificate then select an option to use this custom certificate.

Note

For more information on using custom certificates, refer to "Technical Bulletin 17877: Using Custom Certificates With SoundPoint IP Phones" at www.polycom.com/support/voice/.

Configuration changes can be performed locally:

Local	Local Phone User Interface	The custom certificate can be specified and the type of certificate to trust can be set under the Settings menu.
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Incoming Signaling Validation

The three optional levels of security for validating incoming network signaling are:

- Source IP address validation
- Digest authentication
- Source IP address validation and digest authentication

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: sip.cfg	Specify the type of validation to perform on a request-by-request basis, appropriate to specific event types in some cases. <ul style="list-style-type: none"> For more information, refer to Request Validation <requestValidation/> on page A-14.
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Configuration File Encryption

Configuration files (excluding the master configuration file), contact directories, and configuration override files can all be encrypted.

Note

The SoundPoint IP 300 and 500 phones will always fail at decrypting files. These phones will recognize that a file is encrypted, but cannot decrypt it and will display an error. Encrypted configuration files can only be decrypted on the SoundPoint IP 301, 320, 330, 430, 501, 550, 560, 600, 601, and 650 and the SoundStation IP 4000 phones.

The master configuration file cannot be encrypted on the boot server. This file is downloaded by the bootROM that does not recognize encrypted files. For more information, refer to [Master Configuration Files](#) on page [A-2](#).

For more information on encrypting configuration files including determining whether an encrypted file is the same as an unencrypted file and using the SDK to facilitate key generation, refer to [Encrypting Configuration Files](#) on page [C-3](#).

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: sip.cfg	Specify the phone-specific contact directory and the phone-specific configuration override file. <ul style="list-style-type: none"> For more information, refer to Encryption <encryption/> on page A-80.
	Configuration file: <device>.cfg	Change the encryption key. <ul style="list-style-type: none"> For more information, refer to refer to Flash Parameter Configuration on page A-105.

Configuring SoundPoint IP / SoundStation IP Phones Locally

A local phone-based configuration web server is available, unless it is disabled through **sip.cfg**. It can be used as the only method of modifying phone configuration or as a distributed method of augmenting a centralized provisioning model. For more information, refer to [Web Server <httpd/>](#) on page [A-57](#).

The phone's local user interface also permits many application settings to be modified, such as SIP server address, ring type, or regional settings such as time/date format and language.

Local Web Server Access	<p>Point your web browser to <code>http://<phoneIPAddress>/</code>.</p> <p>Configuration pages are accessible from the menu along the top banner.</p> <p>The web server will issue an authentication challenge to all pages except for the home page.</p> <p>Credentials are (case sensitive):</p> <p>User Name: Polycom</p> <p>Password: The administrator password is used for this.</p>
Local Settings Menu Access	<p>Some items in the Settings menu are locked to prevent accidental changes. To unlock these menus, enter the user or administrator passwords.</p> <p>The administrator password can be used anywhere that the user password is used.</p> <p>Factory default passwords are:</p> <p>User password: 123</p> <p>Administrator password: 456</p>
Passwords:	
Administrator password required.	<p>Network Configuration</p> <p>SIP Configuration</p> <p>SSL Security settings</p> <p>Reset to Default - local configuration, device settings, and file system format</p>
User password required.	Restart Phone

Changes made through the web server or local user interface are stored internally as overrides. These overrides take precedence over settings contained in the configuration obtained from the boot server.

If the boot server permits uploads, these override setting will be saved in a file called `<Ethernet address>-phone.cfg` on the boot server as well in flash memory.

Warning

Local configuration changes will continue to override the boot server-derived configuration until deleted through the Reset Local Config menu selection.

Troubleshooting Your SoundPoint IP / SoundStation IP Phones

This chapter provides you with some tools and techniques for troubleshooting SoundPoint IP / SoundStation IP phones and installations. The phone can provide feedback in the form of on-screen error messages, status indicators, and log files for troubleshooting issues.

This chapter includes information on:

- [BootROM Error Messages](#)
- [Application Error Messages](#)
- [Status Menu](#)
- [Log Files](#)

This chapter also presents phone issues, likely causes, and corrective actions. Issues are grouped as follows:

- [Power and Startup](#)
- [Controls](#)
- [Access to Screens and Systems](#)
- [Calling](#)
- [Displays](#)
- [Audio](#)
- [Upgrading](#)

Review the latest *Release Notes* for the SIP application for known problems and possible workarounds. For the latest *Release Notes* and the latest version of this Administrator's Guide, go to Polycom Technical Support at <http://www.polycom.com/support/voice/>.

If your problem is not listed in this chapter nor described in the latest *Release Notes*, contact your Certified Polycom Reseller for support.

Error Messages

There are several different error messages that can be displayed on the phone when it is booting. Some of these errors are fatal, meaning that the phone will not be able to boot until this issue has been resolved, and some are recoverable, meaning the phone will continue booting after the error, but the configuration of the phone may not be what you were expecting.

BootROM Error Messages

Most of these errors are also logged on the phone's boot log, however, if you are having trouble connecting to the boot server, the phone will likely not be able to upload the boot log for you to examine.

Failed to get boot parameters via DHCP

The phone does not have an IP address and therefore cannot boot. Check that all cables are connected, the DHCP server is running and that the phone has not been put into a VLAN which is different from the DHCP server. Check the DHCP configuration.

Application <file name> is not compatible with this phone!

When the bootROM displays an error like "The application is not compatible", it means an application file was downloaded from the boot server, but it cannot be installed on this phone. This issue can usually be resolved by finding a software image that is compatible with the hardware or the bootROM being used and installing this on the boot server. There are various different hardware and software dependencies. Refer to the latest *Release Notes* for details on the version you are using.

Could not contact boot server, using existing configuration

The phone could not contact the boot server, but the causes may be numerous. It may be a cabling issue, it may be related to DHCP configuration, or it could be a problem with the boot server itself. The phone can recover from this error so long as it previously downloaded a valid application bootROM image and all of the necessary configuration files.

Error, application is not present!

There is no application stored in flash memory and the phone cannot boot. A compatible SIP application must be downloaded into the phone using one of the supported provisioning protocols. You need to resolve the issue of connecting to the boot server. This error is typically a result of one of the above errors. This error is fatal.

Not all configuration files were present on the server

Similarly, a message about configuration files not being present, means that the phone was able to reach the boot server, but that it was not able to find all the necessary files. So long as the files exist in flash memory, the phone can boot following this error.

Note

This error does not occur with the current BootROM.

Error loading <file name>

When the required file does not exist in flash memory and cannot be found on the boot server, the “Error loading” message will tell you which file could not be found. This error only remains on the screen for a few seconds so you need to watch closely. The phone reboots.

Note

This error does not occur with the current BootROM.

Application Error Messages

Config file error. Error is <Hex #>


If there is an error in the configuration file, you will not be able to reboot the phones. You must review the boot server configuration, make the correction, and reapply the configuration file by restarting the phones.

Network link is down

Since the SoundPoint IP / SoundStation IP phones do not have an LED indicating network LINK status like many networking devices, if a link failure is detected while the phone is running a message saying “Network link is down” will be displayed. This message will be shown on the screen whenever the phone is not in the menu system and will remain on screen until the link problem is resolved.

Status

When the phone is unable to register with the call control server, the icon

 is shown (outline). Once the phone is able to register, the icon  is shown (solid).

Blinking Time

If the phone has not been able to contact the SNTP server or if one has not been configured, the date/time display will flash until this is fixed. If an SNTP is not available, the data/time display can be turned off so that the flashing display is not a distraction.

Status Menu

Debugging of single phone may be possible through an examination of the phone's status menu. Press **Menu**, select **Status**, and then press the **Select** soft key.

Under the **Platform** selection, you can get details on the phone's serial number or MAC address, the current IP address, the bootROM version, the application version, the name of the configuration files in use, and the address of the boot server.

In the **Network** menu, the phone will provide information about TCP/IP setting, Ethernet port speed, connectivity status of the PC port, and statistics on packets sent and received since last boot. This would also be a good place to look and see how long it's been since the phone rebooted. The Call Statistics screen shows packets sent and received on the last call.

The **Lines** menu will give you details about the status of each line that has been configured on the phone.

Finally, the **Diagnostics** menu offers a series of hardware tests to verify correct operation of the microphone, speaker, handset, and third party headset, if present. It will also let you test that each of the keys on the phone is working, and it will display the function that has been assigned to each of the keys in the configuration. This is also where you can test the LCD for faulty pixels.

In addition to the hardware tests, the Diagnostics menu has a series of real-time graphs for CPU, network and memory utilization that can be helpful in diagnosing performance issues.

Log Files

SoundPoint IP and SoundStation IP phones will log various events to files stored in the flash file system and will periodically upload these log files to the boot server. The files are stored in the phone's home directory or a user-configurable directory.

There is one log file for the bootROM and one for the application. When a phone uploads its log files, they are saved on the boot server with the MAC address of the phone prepended to the file name. For example, **00f4f200360b-boot.log** and **00f4f200360b-app.log** are the files associated with

MAC address 00f4f200360b. The bootROM log file is uploaded to the boot server after every reboot. The application log file is uploaded periodically or when the local copy reaches a predetermined size.

Both log files can be uploaded on demand using a multiple key combination described in [Multiple Key Combinations](#) on page C-9. The phone uploads four files, namely, **mac-boot.log**, **app-boot.log**, **mac-now-boot.log**, and **mac-now-app.log**. The “now_” logs are uploaded manually.

The amount of logging that the phone performs can be tuned for the application to provide more or less detail on specific components of the phone’s software. For example, if you are troubleshooting a SIP signaling issue, you are not likely interested in DSP events. Logging levels are adjusted in the configuration files or via the web interface. You should not modify the default logging levels unless directed to by Polycom Technical Support. Inappropriate logging levels can cause performance issues on the phone.

In addition to logging events, the phone can be configured to automatically execute command-line instructions at specified intervals that output run-time information such as memory utilization, task status, or network buffer contents to the log file. These techniques should only be used in consultation with Polycom Technical Support.

Application Logging Options

Each of the components of the application software is capable of logging events of different severity. This allows you to capture lower severity events in one part of the application, while still only getting high severity event for other components.

The parameters for log level settings are found in the **sip.cfg** configuration file. They are `log.level.change.module_name`. Log levels range from 1 to 6 (1 for the most detailed logging, 6 for critical errors only). There are currently 27 different log levels that can be adjusted to assist with the investigation of different problems.

When testing is complete, remember to return all logging levels to the default value of 4.

There are other logging parameters that you may wish to modify. Changing these parameters does not have the same impact as changing the logging levels, but you should still understand how your changes will affect the phone and the network.

- `log.render.level` – Sets the lowest level that can be logged (default=1)
- `log.render.file.size` – Maximum size before log file is uploaded (default=16 kb)
- `log.render.file.upload.period` – Frequency of log uploads (default is 172800 seconds = 48 hours)
- `log.render.file.upload.append` – Controls if log files on the boot server are overwritten or appended, not supported by all servers

- `log.render.file.upload.append.sizeLimit`—Controls the maximum size of log files on the boot server (default=512 kb)
- `log.render.file.upload.append.limitMode`—Controls action to take when server log reaches max size, actions are stop and delete

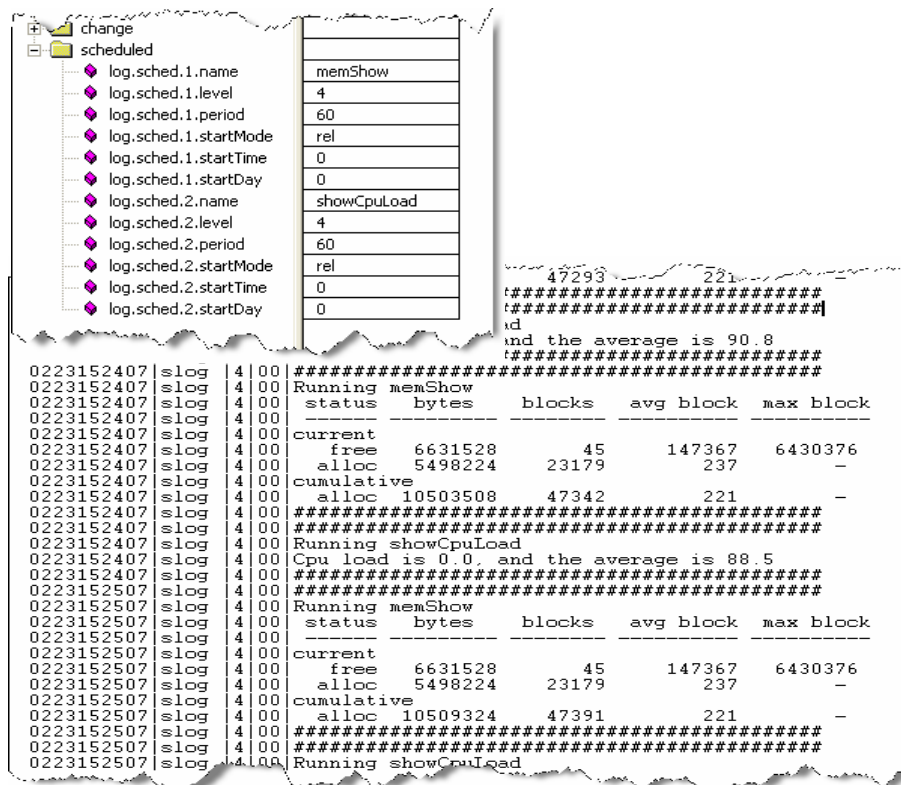
Scheduled Logging

Scheduled logging is a powerful tool for anyone who is trying to troubleshoot an issue with the phone that only occurs after some time in operation.

The output of these instructions is written to the application log, and can be examined later (for trend data).

The parameters for scheduled logging are found in the `sip.cfg` configuration file. They are `log.sched.module_name`.

The following figure shows an example of a configuration file and the resulting log file.



Manual Log Upload

If you want to look at the log files without having to wait for the phone to upload them (which could take as long as 24 hours or more), initiate an upload by pressing correct combination of keys on the phone.

For more information, refer to [Multiple Key Combinations](#) on page C-9.

When the log files are manually uploaded, the word “now” is inserted into the name of the file, for example, 0004f200360b-now-boot.log .

Reading a Boot Log

The following figure shows a portion of a boot log file:

```

0223214053|so|4|00|----- Initial log entry -----
0223214053|so|4|00|++
0223214053|wdog|4|00|In+++ Note that bootrom log times are in GMT
0223214053|lcf|4|00|In
0223214053|4|00|Initial log entry
0223214053|5|00|CDP is DISABLED.
0223214053|5|00|80
0223214053|3|00|P1 802.1Q/VLAN tagging is DISABLED
0223214053|3|00|Platform: Board=2345-11500-020 A
0223214053|3|00|Platform: MAC=0004f2015a51, IP=Unknown, Subnet Mask=Unknown
0223214053|3|00|Platform: BootBlock=2.5.0 (11500_020) 20-Aug-04 16:05
0223214053|3|00|Application, main: Label=BOOT, Version=3.0.1.0026 29-Mar-05 10:29
0223214053|3|00|Application, main: P/N=3150-11069-301
0223214053|4|00|Initial log entry.
0223214054|so|3|00|Link status is Net up Speed 10 half Duplex. PC down.
0223214055|app1|3|00|Using resolver server 172.23.0.200, alternate server 172.23.0.239, and do
0223214123|app1|3|00|Time
0223214123|app1|3|00|DHCP returned result 0x3E7 from server 172.23.0.232
0223214123|app1|3|00|Phone IP address is 172.23.2.172.
0223214123|app1|3|00|Subnet mask is 255.255.0.0.
0223214123|app1|3|00|Gateway address is 172.23.2.240.
0223214123|app1|3|00|Time server is 172.23.0.235.
0223214123|app1|3|00|GMT offset is -28800 seconds.
0223214123|app1|3|00|DNS server is 172.23.0.200.
0223214123|app1|3|00|Bo DNS alternate server is 172.23.0.239.
0223214123|app1|3|00|Bo DNS domain is vancouver.polycom.com.
0223214259|cfg|3|00|Image
0223214300|cfg|3|00|Image sip.ld has not changed.
0223214326|app1|4|00|Loaded application sip.ld successfully. errors 0x0.
0223214326|app1|6|00|Uploading boot log. time is THU FEB 23 21:43:26 2006

```

Boot Failure Messages

The following figure shows an example of “Application sip.ld is not compatible with this phone!” boot failure messages:

```

0223220825|cfg|3|00|New load header information:
0223220825|cfg|3|00|Code length: 0x0013F22D
0223220825|cfg|3|00|Header check Sum: 0x20147429
0223220825|cfg|3|00|Code check Sum: 0x0A8C0D28
0223220825|cfg|3|00|Options: 0x00000003
0223220828|cfg|3|00|New load header information:
0223220828|cfg|3|00|Code length: 0x0013F16D
0223220828|cfg|3|00|Header check Sum: 0x20147369
0223220828|cfg|3|00|Code check Sum: 0x0A8DE14D
0223220828|cfg|3|00|Options: 0x00000003
0223220828|cfg|3|00|Using compatible image 1.
0223220834|sig|4|00|Initial log entry
0223220834|sig|3|00|hwSigParseRemove: could not find key
0223220834|cfg|5|00|Bad image signature.
0223220834|cfg|5|00|Error updating application.
0223220834|app1|4|00|Application sip.ld is not compatible
0223220904|app1|4|00|Loaded application sip.ld successful.
0223220904|app1|6|00|Uploading boot log. time is THU FEB

```

Reading an Application Log

The following figure shows a portion of an application log file:

```

0224000058 |so|*|00|----- Initial log entry -----
0224000058 |so|*|00|Platform: Model=SoundPoint IP 500, Assembly=2345-11500-020 Rev=A
0224000058 |so|*|00|Platform: MAC=0004f2015a51, IP=172.23.2.172, Subnet Mask=255.255.0.0
0224000058 |so|*|00|Platform: BootBlock=2.5.0 (11500_020) 20-Aug-04 16:05
0224000058 |so|*|00|Platform: Bootrom=3.0.1.0026 29-Mar-05 10:29
0224000058 |so|*|00|Application, main: Label=SIP, Version=1.6.5.0043 31-Jan-06 11:16
0224000058 |so|*|00|Application, main: P/N=3150-11530-165
0224000058 |hw|*|00|Initial log entry. Current logging level 4
0224000058 |ares|*|00|Initial log entry. Current logging level 4
0224000058 |dns|*|00|Initial log entry. Current logging level 4
0224000058 |cfg|*|00|Initial log entry. Current logging level 4
0224000058 |dns|*|00|Resolver initialized - added 2 nameservers, and set 'vancouver.polycom.c
0224000058 |log|*|00|Initial log entry. Current logging level 4
0224000058 |curl|*|00|Initial log entry. Current logging level 4
0224000058 |copy|*|00|Initial log entry. Current logging level 3
0224000058 |sec|*|00|Initial log entry. Current logging level 4
0224000058 |copy|3|00|UtilCopyC: curl_easy_perform failed: curlRes: 23, respCode 150
0224000058 |copy|3|00|UtilCopyC: curl error: Curl Error strings have been compiled out.
0224000058 |copy|3|00|UtilCopyC: curl error buffer: Failed writing body.
0224000058 |copy|3|00|UtilCopyC: curl_easy_perform failed: curlRes: 23, respCode 150
0224000058 |copy|3|00|UtilCopyC: curl error: Curl Error strings have been compiled out.
0224000058 |copy|3|00|UtilCopyC: curl error buffer: Failed writing body.
0224000058 |copy|3|00|UtilCopyC: Diff: [error] failed! Reading file /ffs0/sip_cfg_...
0224000058 |so|*|00|CoConfiguration: phone1.cfg,sip.cfg
0224000058 |dns|*|00|Res:
0223160058 |log|*|00|Log render level set to 1
0223160058 |rtos|*|00|Initial log entry. Current logging level 4
0223160058 |rdisk|*|00|Initial log entry. Current logging level 4
0223160058 |res|*|00|Initial log entry. Current logging level 4
0223160058 |httpa|*|00|Initial log entry. Current logging level 4
0223160058 |cdp|*|00|Initial log entry. Current logging level 4
0223160058 |cdp|*|00|CDP is DISABLED.
0223160058 |sys|*|00|Initial log entry. Current logging level 4
0223160058 |wdog|*|00|Initial log entry. Current logging level 4
0223160118 |sec|4|00|utilCryptoConfigFileEncrypted: Could not read file /ffs0/local/local-dir
0223160118 |ssps|*|00|Initial log entry. Current logging level 4
0223160118 |net|*|00|Initial log entry. Current logging level 4
0223160118 |httpd|*|00|Initial log entry. Current logging level 4
0223160118 |key|*|00|Initial log entry. Current logging level 4
0223160118 |ssps|*|00|Application, comp. 1: Label=PolyDSP Orion Mem2 FS1, Version=1.3.3.0010 0
0223160118 |ssps|*|00|Application, comp. 1: P/N=3150-11580-133.
0223160118 |pps|*|00|Initial log entry. Current logging level 4
0223160118 |sip|4|00|[ResFinderC]: Failed to download file SoundPointIPWelcome.wav, errno 0xd
0223160121 |so|4|00|[SoToneC]: Failed to find tone SoundPointIPWelcome.wav - using default
0223160121 |slog|4|00|[SoToneC]: Failed to find tone SoundPointIPWelcome.wav - using default
0223160121 |res|4|00|[ResFinderC]: Failed to download file SoundPointIPWelcome.wav, errno 0xd
0223160121 |so|4|00|[SoToneC]: Failed to find tone SoundPointIPWelcome.wav - using default
0223160153 |sip|4|00|Res:
Registration failed User: 2125551212, Error Code:404 Not Fo

```


Power and Startup

Symptom	Problem	Corrective Action
There are power issues.	The SoundPoint IP / SoundStation IP family SIP phone has no power.	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Verify that no lights appear on the unit when it is powered up. • Check if the phone is properly plugged into a functional AC outlet. • Make sure that the phone isn't plugged into a plug controlled by a light switch that is off. • If plugged into a power strip, try plugging directly into a wall outlet instead. • Try the phone in another room where the electricity is known to be working on a particular outlet. • If using PoE, the power supply voltage may be too high or too low.

Controls

Symptom	Problem	Corrective Action
<p>The dial pad does not work.</p>	<p>The dial pad on the SoundPoint IP / SoundStation IP family SIP phone does not respond.</p>	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Check for a response from other feature keys or from the dial pad. • Place a call to the phone from a known working telephone. Check for display updates. • Press the Menu key followed by System Status and Server Status to check if the telephone is correctly registered to the server. • Press the Menu key followed by System Status and Network Statistics. Scroll down to see if LAN port shows active or Inactive. • Check the termination at the switch or hub end of the network LAN cable. Ensure that the switch/hub port connected to the telephone is operational (if not accessible, contact your system administrator). • Before restarting your phone, contact your system administrator, since this may allow more detailed troubleshooting to occur before losing any current status information.

Access to Screens and Systems

Symptom	Problem	Corrective Action
There is no response from feature key presses.	The SoundPoint IP / SoundStation IP family SIP phone is not in active state.	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Press the keys more slowly. • Check to see whether or not the key has been mapped to a different function or disabled. • Make a call to the phone to check for inbound call display and ringing as normal. If successful, try to press feature keys within the call to access Directory or Buddy Status, for example. • Press Menu followed by Status > Lines to confirm line is actively registered to the call server. • Reboot the phone to attempt re- to the call server (refer to Rebooting the Phone on page C-9).
The display shows “Network Link is Down”.	The LAN cable is not properly connected.	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Check termination at the switch or hub (furthest end of the cable from the phone). • Check that the switch or hub is operational (flashing link/status lights) or contact your system administrator. • Press Menu followed by Status > Network. Scroll down to verify that the LAN is active. • Ping phone from another machine. • Reboot the phone to attempt re- to the call server (refer to Rebooting the Phone on page C-9).

Calling

Symptom	Problem	Corrective Action
There is no dial tone.	Power is not correctly applied to the SoundPoint IP family SIP phone.	Do one of the following: <ul style="list-style-type: none"> • Check that the display is illuminated. • Make sure the LAN cable is inserted properly at the rear of the phone (try unplugging and re-inserting the cable). • If using in-line powering, have your system administrator check that the switch is supplying power to the phone.
	Dial tone is not present on one of audio paths.	Do one of the following: <ul style="list-style-type: none"> • Switch between Handset, Headset (if present) or Hands-Free Speakerphone to see if dial tone is present on another paths. • If dial tone exists on another path, connect a different handset or headset to isolate the problem. • Check configuration for gain levels.
	The phone is not registered.	Contact your system administrator.
The phone does not ring.	Ring setting or volume is low.	Do one of the following: <ul style="list-style-type: none"> • Adjust the ringing level from the front panel using the volume up/down keys. • Check same status of handset, headset (if connected) and through the Hands-Free Speakerphone.
	Outbound or inbound calling is unsuccessful.	Do one of the following: <ul style="list-style-type: none"> • Place a call to the phone under investigation. Check that the display indicates incoming call information. • Lift the handset. Ensure dial tone is present and place a call to another extension or number. Check that the display changes in response.
The line icon shows an unregistered line icon.	The phone line is unregistered.	Contact your system administrator.

Displays

Symptom	Problem	Corrective Action
<p>There is no display.</p> <p>The display is incorrect.</p> <p>The display has bad contrast.</p>	<p>Power is not correctly applied to the SoundPoint IP family SIP phone.</p>	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Check that the display is illuminated. • Make sure the LAN cable is inserted properly at the rear of the phone (try unplugging and re-inserting the cable). • If using in-line powering, have your system administrator check that the switch is supplying power to the phone.
	<p>The contrast needs adjustment.</p>	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Refer to the appropriate SoundPoint IP / SoundStation IP SIP phone User Guide. • Reboot the phone to obtain a default level of contrast (refer to Rebooting the Phone on page C-9).
	<p>Outbound or inbound calling is unsuccessful.</p>	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Place a call to the phone under investigation. Check that the display indicates incoming call information. • Lift the handset. Ensure dial tone is present and place a call to another extension or number. Check that the display changes in response.
<p>The display is flickering.</p>	<p>Certain type of older fluorescent lighting causes the display to appear to flicker.</p>	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Move the SoundPoint IP / SoundStation IP SIP phone away from the lights. • Replace the lights.

Audio

Symptom	Problem	Corrective Action
There is no audio on the headset.	The connections are not correct.	Do one of the following: <ul style="list-style-type: none"> Ensure the headset is plugged into the jack marked Headset at the rear of the phone. Ensure the headset amplifier (if present) is turned on and/or the volume is correctly adjusted).
There are audio and echo issues on the headset.	Possible issues include: <ul style="list-style-type: none"> Echo on external calls through a gateway. Internal calls (no gateway), handsfree echo. Internal calls (no gateway), handset to handset echo. 	Refer to "Technical Bulletin 16249: Troubleshooting Audio and Echo Issues on SoundPoint® IP Phones" on the Polycom Support Knowledgebase.

Upgrading

Symptom	Problem	Corrective Action
SoundPoint IP 300 and/or 500 behave incorrectly or do not display new features.	New features are not supported on the SoundPoint IP 300 and 500 and the configuration files have not been correctly modified. The SoundPoint IP 300 and 500 will not 'understand' the new configuration parameters, and will attempt to load the new application.	The attempt to load the new application will fail since there is no 300/500 image contained within the sip.ld file, so the phone will continue on and run the current version of application that it has in memory. It will however use the new configuration files. Refer to Supporting SoundPoint IP 300 and 500 Phones on page 3-19.

Configuration Files

This appendix provides detailed descriptions of certain configuration files used by the Session Initiation Protocol (SIP) application. It is a reference for all parameters that are configurable when using the centralized provisioning installation model.

This appendix contains information on:

- [Master Configuration Files \(MAC-address.cfg or 000000000000.cfg\)](#)
- [Application Configuration \(sip.cfg\)](#)
- [Per-Phone Configuration \(phone1.cfg\)](#)
- [Flash Parameter Configuration](#)

The application configuration files dictate the behavior of the phone once it is running the executable specified in the master configuration file.

Caution

Configuration files should only be modified by a knowledgeable system administrator. Applying incorrect parameters may render the phone unusable. The configuration files which accompany a specific release of the SIP software must be used together with that software. Failure to do this may render the phone unusable.

Note

In the tables in the following sections, “Null” should be interpreted as the empty string, that is, attributeName="" when the file is viewed in an XML editor.

To enter special characters in a configuration file, enter the appropriate sequence using an XML editor:

- & as &
- " as "
- ' as '
- < as <
- > as >

Note

The various .hd. parameters in **sip.cfg** (such as `voice.aec.hd.enable`, `voice.ns.hd.enable`, and `voice.agc.hd.enable`) are headset parameters. They are not connected to high definition or HD voice.

Master Configuration Files

The master configuration files can be one of:

- Specified master configuration file – The master configuration file can be explicitly specified in the boot server address, for example, `http://usr:pwd@server/dir/example1.cfg`. The filename must end with `.cfg` and be at least five characters long. If this file cannot be downloaded, the phone will search for the per-phone master configuration file (described next).
- Per-phone master configuration file – If per-phone customization is required, the file should be named `<Ethernet address>.cfg`, where Ethernet address is the MAC address of the phone in question. For A-F hexadecimal digits, use upper or lower case, for example, `0004f200106c.cfg`. The Ethernet address can be viewed using the **About** soft key during the auto-restart countdown of the bootROM or through the *Menu > Status > Platform > Phone* menu in the application. It is also printed on a label on the back of the phone. If this file cannot be downloaded, the phone will search for the default master configuration file (described next).
- Default master configuration file – For systems in which the configuration is identical for all phones (no per-phone `<Ethernet address>.cfg` files), the default master configuration file may be used to set the configuration for all phones. The file named `000000000000.cfg` (<12 zeros>.cfg) is the default master configuration file and it is recommended that one be present on the boot server. If a phone does not find its own `<Ethernet address>.cfg` file, it will use this one, and establish a baseline configuration. This file is part of the standard Polycom distribution of configuration files. It should be used as the template for the `<Ethernet address>.cfg` files.

The default master configuration file, `000000000000.cfg`, is shown below:

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File -->
<!-- edit and rename this file to <Ethernet-address>.cfg for
each phone. -->
<!-- $Revision: 1.14 $ $Date 2005/07/27 18:43:30 $ -->
< APPLICATION APP_FILE_PATH="sip.ld"
CONFIG_FILES="phone1.cfg, sip.cfg" MISC_FILES=""
LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY="" />
```

Master configuration files contain six XML attributes:

- `APP_FILE_PATH` – The path name of the application executable. It can have a maximum length of 255 characters. This can be a URL with its own protocol, user name and password, for example `http://usr:pwd@server/dir/sip.ld`.

- **CONFIG_FILES**— A comma-separated list of configuration files. Each file name has a maximum length of 255 characters and the list of file names has a maximum length of 2047 characters, including commas and white space. Each configuration file can be specified as a URL with its own protocol, user name and password, for example
ftp://usr:pwd@server/dir/phone2034.cfg.
- **MISC_FILES**— A comma-separated list of other required files. Dictionary resource files listed here will be stored in the phone's flash file system. So if the phone reboots at a time when the boot server is unavailable, it will still be able to load the preferred language.

Note

On the SoundPoint IP 500, there is insufficient room for a language file. Specifying one will cause a reboot loop

- **LOG_FILE_DIRECTORY**— An alternative directory to use for log files if required. A URL can also be specified. This is blank by default.
- **CONTACTS_DIRECTORY**— An alternative directory to use for user directory files if required. A URL can also be specified. This is blank by default.
- **OVERRIDES_DIRECTORY**— An alternative directory to use for configuration overrides files if required. A URL can also be specified. This is blank by default.
- **LICENSE_DIRECTORY**— An alternative directory to use for license files if required. A URL can also be specified. This is blank by default.

Warning

The order of the configuration files listed in **CONFIG_FILES** is significant:

- The files are processed in the order listed (left to right).
- The same parameters may be included in more than one file.
- The parameter found first in the list of files will be the one that is effective.

This provides a convenient means of overriding the behavior of one or more phones without changing the baseline configuration files for an entire system.

For more information, refer to the “Configuration File Management on SoundPoint IP Phones” whitepaper at www.polycom.com/support/voice/

If you have a requirement for different application loads on different phones on the same boot server, you can create a variable in the master configuration file that is replaced by the MAC address of each phone when it reboots. An example is shown below:

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File -->
<!-- edit and rename this file to <Ethernet-address>.cfg for
each phone. -->
<!-- $RCSfile: 000000000000.cfg,v $ $Revision:$ -->
< APPLICATION APP_FILE_PATH="sip[MACADDRESS].ld"
```

```
CONFIG_FILES="phone1[MACADDRESS].cfg, sip.cfg" MISC_FILES=""
LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY="" />
```

If you have a requirement for separate application loads on different phones on the same boot server, you can modify the application that is loaded when each phone reboots. An example is below:

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File -->
<!-- edit and rename this file to <Ethernet-address>.cfg for
each phone. -->
<!-- $RCSfile: 000000000000.cfg,v $ $Revision:$ -->
< APPLICATION APP_FILE_PATH="sip[PHONE_PART_NUMBER].ld"
CONFIG_FILES="phone1.cfg, sip.cfg" MISC_FILES=""
LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY="" />
```

You can also use the substitution strings PHONE_MODEL, PHONE_PART_NUMBER, and PHONE_MAC_ADDRESS in the master configuration file.

You can also direct phone upgrades to a software image and configuration files based on the phone model number and part number. All XML attributes can be modified in this manner. An example is below:

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File -->
<!-- edit and rename this file to <Ethernet-address>.cfg for
each phone. -->
<!-- $RCSfile: 000000000000.cfg,v $ $Revision:$ -->
<APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="phone1.cfg,
sip.cfg" MISC_FILES="" LOG_FILE_DIRECTORY=""
OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY=""
APP_FILE_PATH_SPIP300="SPIP300.sip.ld"
CONFIG_FILES_SPIP300="phone1_SPIP300.cfg, sip_SPIP300.cfg"
APP_FILE_PATH_SPIP500="SPIP500.sip.ld"
CONFIG_FILES_SPIP500="phone1_SPIP500.cfg, sip_SPIP500.cfg" />
```

For more information, refer to “Technical Bulletin 35361: Overriding Parameters in Master Configuration File on SoundPoint IP Phones” at <http://www.polycom.com/support/voice/>.

Application Configuration

The configuration file **sip.cfg** contains SIP protocol and core configuration settings that would typically apply to an entire installation and must be set before the phones will be operational, unless changed through the local web

server interface or local menu settings on the phone. These settings include the local port used for SIP signaling, the address and ports of a cluster of SIP application servers, voice codecs, gains, and tones, and other parameters.

These parameters include:

- Protocol <volpProt/>
- Dial Plan <dialplan/>
- Localization <lcl/>
- User Preferences <up/>
- Tones <tones/>
- Sampled Audio for Sound Effects <saf/>
- Sound Effects <se/>
- Voice Settings <voice/>
- Quality of Service <QOS/>
- Basic TCP/IP <TCP_IP/>
- Web Server <httpd/>
- Call Handling Configuration <call/>
- Directory <dir/>
- Presence <pres/>
- Fonts
- Keys <key/>
- Backgrounds <bg/>
- Bitmaps <bitmap/>
- Indicators <ind/>
- Event Logging <log/>
- Security <sec/>
- License <license/>
- Provisioning <prov/>
- RAM Disk <ramdisk/>
- Request <request/>
- Feature <feature/>

- [Resource <res/>](#)
- [Microbrowser <mb/>](#)



Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first. For more information, refer to the "Configuration File Management on SoundPoint IP Phones" whitepaper at www.polycom.com/support/voice/.

Protocol <volpProt/>

This attribute includes:

- [Local <local/>](#)
- [Server <server/>](#)
- [SDP <SDP/>](#)
- [SIP <SIP/>](#)

Local <local/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.local.port	0 to 65535	5060	Local port for sending and receiving SIP signaling packets. If set to 0 or Null, 5060 is used for the local port but it is not advertised in the SIP signaling. If set to some other value, that value is used for the local port and it is advertised in the SIP signaling.

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.server.dhcp.available	0, 1	0	If set to 1, check with the DHCP server for SIP server IP address. If set to 0, do not check with DHCP server.
volpProt.server.dhcp.option	128 to 255	Null	Option to request from the DHCP server if volpProt.server.dhcp.available = 1. There is no default value for this parameter, it must be filled in with a valid value. Note: If the reg.x.server.y.address parameter in Registration <reg/> on page A-89 is non-Null, it takes precedence even if the DHCP server is available.
volpProt.server.dhcp.type	0, 1	Null	If set to 0, IP request address. If set to 1, request string. Type to request from the DHCP server if volpProt.server.dhcp.available = 1. There is no default value for this parameter, it must be filled in with a valid value.
volpProt.server.x.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that accepts registrations. Multiple servers can be listed starting with x=1, 2, ... for fault tolerance.
volpProt.server.x.port	0, Null, 1 to 65535	Null	Note: If the reg.x.server.y.address parameter in Registration <reg/> on page A-89 is non-Null, <u>all</u> of the reg.x.server.y.xxx parameters will override the volpProt.server parameters. If port is 0 or Null: If voIpProt.server.x.address is a hostname and voIpProt.server.x.transport is set to DNSNaptr, do NAPTR then SRV lookups. If voIpProt.server.x.transport is set to TCPpreferred or UDPOnly then use 5060 and don't advertise the port number in signalling. If voIpProt.server.x.address is an IP address, there is no DNS lookup and 5060 is used for the port but it is not advertised in signalling. If port is 1 to 65535: This value is used and it is advertised in signalling.

Attribute	Permitted Values	Default	Interpretation
volpProt.server.x.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	<p>If set to Null or DNSnaptr: If voIpProt.server.x.address is a hostname and volpProt.server.x.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If voIpProt.server.x.address is an IP address, or a port is given, then UDP is used.</p> <p>If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails.</p> <p>If set to UDPOnly: Only UDP will be used.</p> <p>If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.</p> <p>If set to TCPOnly: Only TCP will be used.</p> <p>NOTE: TLS is not supported on SoundPoint IP 300 and 500 phones.</p>
volpProt.server.x.expires	positive integer, minimum 300	3600	<p>The phone's requested registration period in seconds.</p> <p>Note: The period negotiated with the server may be different. The phone will attempt to re-register at the beginning of the overlap period. For example, if "expires"=3600 and "overlap"=60, the phone will re-register after 3540 seconds (3600 – 60).</p>
volpProt.server.x.expires.overlap	positive integer, minimum 5, maximum 65535	60	<p>The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone will try to re-register at half the expiration time returned by the server if that value is less than the configured overlap value.</p>
volpProt.server.x.register	0, 1	1	<p>If set to 0, calls can be routed to an outbound proxy without registration.</p>
volpProt.server.x.retryTimeOut	Null or non-negative integer	0	<p>If set to 0 or Null, use standard RFC 3261 signaling retry behavior. Otherwise retryTimeOut determines how often retries will be sent.</p> <p>Units = milliseconds. (Finest resolution = 100ms).</p>

Attribute	Permitted Values	Default	Interpretation
volpProt.server.x.retryMaxCount	Null or non-negative integer	3	If set to 0 or Null, 3 is used. retryMaxCount retries will be attempted before moving on to the next available server.
volpProt.server.x.expires.lineSeize	positive integer, minimum 10	30	Requested line-seize subscription period.
volpProt.server.x.lcs	0, 1	0	This attribute overrides the voIpProt.SIP.lcs . If set to 1, the proprietary "epid" parameter is added to the From field of all requests to support Microsoft Live Communications Server.

SDP <SDP/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SDP.answer.useLocalPreferences	0 or 1	0	If set to 1, the phones uses its own preference list when deciding which codec to use rather than the preference list in the offer. If set to 0, it is disabled.

SIP <SIP/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.useContactInReferTo	0, 1	0	If set to 0, the "To URI" is used in the REFER. If set to 1, the "Contact URI" is used in the REFER.
volpProt.SIP.useRFC2543hold	0, 1	0	If set to 1, use the obsolete c=0.0.0.0 RFC2543 technique, otherwise, use SDP media direction attributes (such as a=sendonly) per RFC 3264 when initiating hold. In either case, the phone processes incoming hold signaling in either format.
volpProt.SIP.useSendonlyHold	0, 1	1	If set to 1, the phone will send a reinvite with a stream mode attribute of "sendonly" when a call is put on hold. This is the same as the previous behavior. If set to 0, the phone will send a reinvite with a stream mode attribute of "inactive" when a call is put on hold. NOTE: The phone will ignore the value of this parameter if set to 1 when the parameter voIpProt.SIP.useRFC2543hold is also set to 1 (default is 0).
volpProt.SIP.lcs	0, 1	0	If set to 1, the proprietary "epid" parameter is added to the From field of all requests to support Microsoft Live Communications Server.
volpProt.SIP.ms-forking	0, 1	0	If set to 0, support for MS-forking is disabled. If set to 1, support for MS-forking is enabled and the phone will reject all Instant Message INVITEs. This parameter is relevant for Microsoft Live Communications Server server installations. Note that if any end point registered to the same account has MS-forking disabled, all other end points default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the end points is Windows Messenger.
volpProt.SIP.dialog.usePvalue	0, 1	0	If set to 0, phone uses "pval" field name in Dialog. This obeys the draft-ietf-sipping-dialog-package-06.txt draft. If set to 1, phone uses a field name of "pvalue".

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.connectionReuse.useAlias	0, 1	0	If set to 0, this is the old behavior. If set to 1, phone uses the connection reuse draft which introduces "alias".
volpProt.SIP.sendCompactHdrs	0, 1	0	If set to 0, SIP header names generated by the phone use the long form, for example 'From'. If set to 1, SIP header names generated by the phone use the short form, for example 'f'.
volpProt.SIP.keepalive.sessionTimers	0, 1	0	If set to 1, the session timer will be enabled. If set to 0, the session timer will be disabled, and the phone will not declare "timer" in "Support" header in INVITE. The phone will still respond to a re-INVITE or UPDATE. The phone will not try to re-INVITE or do UPDATE even if remote end point asks for it.
volpProt.SIP.requestURI.E164.addGlobalPrefix	0, 1	0	If set to 1, '+' global prefix is added to E.164 user parts in sip: URIs:.
volpProt.SIP.allowTransferOnProceeding	0, 1	1	If set to 1, a transfer can be completed during the proceeding state of a consultation call. If set to 0, a transfer is not allowed during the proceeding state of a consultation call. If set to Null, the default value is used.
volpProt.SIP.dialog.useSDP	0, 1	0	If set to 0, new dialog event package draft is used (no SDP in dialog body). If set to 1, for backwards compatibility, use this setting to send SDP in dialog body.
volpProt.SIP.pingInterval	0 to 3600	0	The number in seconds to send "PING" message. This feature is disabled by default.
volpProt.SIP.useContactInReferTo	0, 1	0	If set to 1, the Contact URI is used. If set to 0, the TO URI is used (previous behavior).
volpProt.SIP.serverFeatureControl.cf	0, 1	0	If set to 1, server-based call forwarding is enabled. The call server has control of call forwarding. If set to 0, server-based call forwarding is not enabled. This is the old behavior.

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.serverFeatureControl.dnd	0, 1	0	If set to 1, server-based DND is enabled. The call server has control of DND. If set to 0, server-based DND is not enabled. This is the old behavior.
volpProt.SIP.authOptimizedInFailover	0,1	0	If set to 1, when failover occurs, the first new SIP request is sent to the server that sent the proxy authentication request. If set to 0, when failover occurs, the first new SIP request is sent to the server with the highest priority in the server list. If reg.x.auth.optimizedInFailover set to Null, this attribute is checked. If voIpProt.SIP.authOptimizedInFailover is Null, then this feature is disabled. If both attributes are set, the value of reg.x.auth.optimizedInFailover takes precedence.
volpProt.SIP.csta	0, 1	0	If set to 1, uaCSTA is enabled.

This attribute also includes:

- [Outbound Proxy <outboundProxy/>](#)
- [Alert Information <alertInfo/>](#)
- [Request Validation <requestValidation/>](#)
- [Special Events <specialEvent/>](#)
- [Conference Setup <conference/>](#)

Outbound Proxy <outboundProxy/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.outboundProxy.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server to which the phone shall send all requests.
volpProt.SIP.outboundProxy.port	1 to 65535	5060	
volpProt.SIP.outboundProxy.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	<p>If set to Null or DNSnaptr: If volpProt.SIP.outboundProxy.address is a hostname and voIpProt.SIP.outboundProxy.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If voIpProt.SIP.outboundProxy.address is an IP address, or a port is given, then UDP is used.</p> <p>If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails.</p> <p>If set to UDPOnly: Only UDP will be used.</p> <p>If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.</p> <p>If set to TCPOnly: Only TCP will be used.</p> <p>NOTE: TLS is not supported on SoundPoint IP 300 and 500 phones.</p>

Alert Information <alertInfo/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.alertInfo.x.value	string to compare against the value of Alert-Info headers in INVITE requests	Null	Alert-Info fields from INVITE requests will be compared against as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class (refer to Ring type <rt/> on page A-33) will be applied.
volpProt.SIP.alertInfo.x.class	positive integer	Null	

Request Validation <requestValidation/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.requestValidation.x.request	One of: "INVITE", "ACK", "BYE", "REGISTER", "CANCEL", "OPTIONS", "INFO", "MESSAGE", "SUBSCRIBE", "NOTIFY", "REFER", "PRACK", or "UPDATE"	Null	Sets the name of the method for which validation will be applied. WARNING: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases, therefore, use it wisely.
volpProt.SIP.requestValidation.x.method	Null or one of: "source", "digest" or "both"/"all"	Null	If Null, no validation is done. Otherwise this sets the type of validation performed for the request: <i>source</i> : ensure request is received from an IP address of a server belonging to the set of target registration servers; <i>digest</i> : challenge requests with digest authentication using the local credentials for the associated registration (line); <i>both</i> or <i>all</i> : apply both of the above methods

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.requestValidation.x.request.y.event	A valid string	Null	Determines which events specified with the Event header should be validated; only applicable when <code>volpProt.SIP.requestValidation.x.request</code> is set to "SUBSCRIBE" or "NOTIFY". If set to Null, all events will be validated.
volpProt.SIP.requestValidation.digest.realm	A valid string	Polycorn SPIP	Determines string used for Realm.

Special Events <specialEvent/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.specialEvent.lineSeize.nonStandard	0, 1	1	If set to 1, process a 200 OK response for a line-seize event SUBSCRIBE as though a line-seize NOTIFY with Subscription State: active header had been received, this speeds up processing.
volpProt.SIP.specialEvent.checkSync.alwaysReboot	0, 1	0	If set to 1, always reboot when a NOTIFY message is received from the server with event equal to check-sync. If set to 0, only reboot if any of the files listed in <MAC-address>.cfg have changed on the FTP server when a NOTIFY message is received from the server with event equal to check-sync.

Conference Setup <conference/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SIP.conference.address	ASCII string up to 128 characters long	Null	If Null, conferences are set up on the phone locally. If set to some value, conferences are set up by the server using the conferencing agent specified by this address. The acceptable values depend on the conferencing server implementation policy.

Dial Plan <dialplan/>

Note

The dial plan is not applied against Placed Call List, VoiceMail, last call return, and remote control dialed numbers.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.applyToCallListDial	0, 1	0	This attribute covers dialing from Received Call List and Missed Call List including dialing from Edit or Info sub- menus. If set to 0, the dial plan is not applied against the dialed number. if set to 1, the dial plan is applied against the dialed number.
dialplan.applyToDirectoryDial	0, 1	0	This attribute covers dialing from Directory as well as Speed Dial List. Value interpretation is the same as for dialplan.applyToCallListDial. Note: An Auto Call Contact number is considered a dial from directory.
dialplan.applyToUserDial	0, 1	1	This attribute covers the case when the user presses the Dial soft key to send dialed number when in idle state display. Value interpretation is the same as for dialplan.applyToCallListDial.
dialplan.applyToUserSend	0, 1	1	This attribute covers the case when the user presses the Send soft key to send the dialed number. Value interpretation is the same as for dialplan.applyToCallListDial.
dialplan.impossibleMatchHandling	0, 1 or 2	0	If set to 0, the digits entered up to and including the point where an impossible match occurred are sent to the server immediately. If set to 1, give reorder tone. If set to 2, allow user to accumulate digits and dispatch call manually with the Send soft key.
dialplan.removeEndOfDial	0, 1	1	If set to 1, strip trailing # digit from digits sent out.

This attributes also includes:

- [Digit Map <digitmap/>](#)
- [Routing <routing/>](#)

Digit Map <digitmap/>

A digit map is defined either by a “string” or by a list of strings. Each string in the list is an alternative numbering scheme, specified either as a set of digits or timers, or as an expression over which the gateway will attempt to find a shortest possible match.

Digit map extension letter “R” indicates that certain matched strings are replaced. The following examples shows the semantics of the syntax:

- R9RRxxxxxxx – Remove 9 at the beginning of the dialed number
 - For example, if a customer dials 914539400, the first 9 is removed when the call is placed.
- RR604Rxxxxxxx – Prepend 604 to all 7 digit numbers
 - For example, if a customer dials 4539400, 604 is added to the front of the number, so a call 6044539400 is placed.
- R9R604Rxxxxxxx – Replaces 9 with 604
- R999R911R – Convert 999 to 911
- xxR601R600Rxx – When applied on 1160122 gives 1160022
- xR60xR600Rxxxxxxx – Any 60x will be replaced with 600 in the middle of the dialed number that matches
 - For example, if a customer dials 16092345678, a call is placed to 16002345678.
- 911xxx.T – A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct
 - For example:
 - 91112 with waiting time to comply with T is a match
 - 911123 with waiting time to comply with T is a match
 - 9111234 with waiting time to comply with T is a match
 - and the number can grow indefinitely given that pressing the next digit takes less than T.

The following guidelines should be noted:

- You must use only *, #, +, or 0-9 between second and third R
- If a digit map does not comply, it is not included in the digit plan as a valid one. That is, no matching is done against it.

- There is no limitation on the number of R triplet sets in a digit map. However, a digit map that contains less than full number of triplet sets (for example, a total of 2Rs or 5Rs) is considered an invalid digit map.
- Using T in the left part of RRR syntax is not recommended. For example, R0TR322R should be avoided.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.digitmap	string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. String is limited to 768 bytes and 30 segments; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on; '+' is allowed as a valid digit; extension letter 'R' is used as defined above.	[2-9]11 0T +011xxx.T 0[2-9]xxxxxxxx +1[2-9]xxxxxxxx [2-9]xxxxxxxx [2-9]xxxT	When this attribute is present, number-only dialing during the setup phase of new calls will be compared against the patterns therein and if a match is found, the call will be initiated automatically eliminating the need to press Send. Attributes dialplan.applyToCallListDial, dialplan.applyToDirectoryDial, dialplan.applyToUserDial, and dialplan.applyToUserSend control the use of match and replace in the dialed number in the different scenarios.
dialplan.digitmap.timeOut	string of positive integers separated by ' '	3 3 3 3 3 3	Timeout in seconds for each segment of digit map. Note: If there are more digit maps than timeout values, the default value of 3 will be used. If there are more timeout values than digit maps, the extra timeout values are ignored.

Routing <routing/>

This attribute allows the user to create a specific routing path for outgoing SIP calls independent of other "default" configurations.

This attribute also includes:

- [Server <server/>](#)
- [Emergency <emergency/>](#)

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.routing.server.x.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that will be used for routing calls. Multiple servers can be listed starting with x=1, 2, ... for fault tolerance.
dialplan.routing.server.x.port	1 to 65535	5060	

Emergency <emergency/>

In the following attributes, x is the index of the emergency entry description and y is the index of the server associated with emergency entry x . For each emergency entry (index x), one or more server entries (indexes (x,y)) can be configured. x and y must both use sequential numbering starting at 1.

Attribute	Permitted Values	Default	Interpretation
dialplan.routing.emergency.x.value	Single entry representing a SIP URL	for $x = 1$, value = "911", Null for all others	This determines the URLs that should be watched for. When one of these defined URLs is detected as having been dialed by the user, the call will automatically be directed to the defined emergency server.
dialplan.routing.emergency.x.server.y	positive integer	for $x=1, y = 1$, Null for all others	Index representing the server defined in Server <server/> on page A-19 that will be used for emergency routing.

Localization <lcl/>

The phone has a multilingual user interface. It supports both North American and international time and date formats. The call progress tones can also be customized. For more information, refer to [Chord-Sets <chord/>](#) on page A-26, and [Call Progress Patterns](#) on page A-30.

This attribute includes:

- [Multilingual <ml/>](#)
- [Date and Time <datetime/>](#)

Multilingual <ml/>

The multilingual feature is based on string dictionary files downloaded from the boot server. These files are encoded in standalone XML format. Several western European and Asian languages are included with the distribution.

Attribute	Permitted Values	Interpretation
lcl.ml.lang	Null OR An exact match for one of the folder names under the SoundPointIPLocalization folder on the boot server.	If Null, the default internal language (US English) will be used, otherwise, the language to be used may be specified in the format <i>language-region</i> .
lcl.ml.lang.menu.x	String in the format <i>language_region</i>	Multiple lcl.ml.lang.menu.x attributes are supported - as many languages as are desired. However, the lcl.ml.lang.menu.x attributes must be sequential (lcl.ml.lang.menu.1, lcl.ml.lang.menu.2, lcl.ml.lang.menu.3, ..., lcl.ml.lang.menu.N) with no gaps and the strings must exactly match a folder name under the SoundPointIPLocalization folder on the boot server for the phone to be able to locate the dictionary file.
lcl.ml.lang.clock.x.24HourClock	0,1	If attribute present, overrides lcl.datetime.time.24HourClock. If 1, display time in 24-hour clock mode rather than am/pm.
lcl.ml.lang.clock.x.format	string which includes 'D', 'd' and 'M' and two optional commas	If attribute present, overrides lcl.datetime.date.format; D = day of week d = day M = month Up to two commas may be included. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: "D,,dM" is illegal.

Attribute	Permitted Values	Interpretation
lcl.ml.lang.clock.x.longFormat	0, 1	If attribute present, overrides lcl.datetime.date.longFormat. If 1, display the day and month in long format (Friday/November), otherwise use abbreviations (Fri/Nov).
lcl.ml.lang.clock.x.dateTop	0, 1	If attribute present, overrides lcl.datetime.date.dateTop. If 1, display date above time, otherwise display time above date.
lcl.ml.lang.y.list	"All" or a comma-separated list	A list of the languages supported on hardware platform 'y' where 'y' can be IP_500 or IP_600.

To add new languages to those included with the distribution:

1. Create a new dictionary file based on an existing one.
2. Change the strings making sure to encode the XML file in UTF-8 but also ensuring the UTF-8 characters chosen are within the Unicode character ranges indicated in the tables below.
3. Place the file in an appropriately named folder according to the format *language_region* parallel to the other dictionary files under the SoundPointIPLocalization folder on the boot server.
4. Add a lcl.ml.lang.clock.menu.x attribute to the configuration file.
5. Add lcl.ml.lang.clock.x.24HourClock, lcl.ml.lang.clock.x.format, lcl.ml.lang.clock.x.longFormat and lcl.ml.lang.clock.x.dateTop attributes and set them according to the regional preferences.
6. (Optional) Set lcl.ml.lang to be the new *language_region* string.

Basic character support includes the following Unicode character ranges	
Name	Range
C0 Controls and Basic Latin	U+0000 - U+007F
C1 Controls and Latin-1 Supplement	U+0080 - U+00FF
Cyrillic (partial)	U+0400 - U+045F

Extended character support available on SoundPoint IP 600 and SoundStation IP 4000 platforms includes the following Unicode character ranges	
Name	Range
CJK Symbols and Punctuation	U+3000 - U+303F
Hiragana	U+3040 - U+309F
Katakana	U+30A0 - U+30FF
Bopomofo	U+3100 - U+312F
Hangul Compatibility Jamo	U+3130 - U+318F
Bopomofo Extended	U+31A0 - U+31BF
Enclosed CJK Letters and Months	U+3200 - U+327F
CJK Compatibility	U+3300 - U+33FF
CJK Unified Ideographs	U+4E00 - U+9FFF
Hangul Syllables	U+AC00 - U+D7A3
CJK Compatibility Ideographs	U+F900 - U+FAFF
CJK Half-width forms	U+FF00 - U+FFFF

Note

Within a Unicode range, some characters may not be supported due to their infrequent usage

Date and Time <datetime/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Interpretation
lcl.datetime.time.24HourClock	0,1	If 1, display time in 24-hour clock mode rather than a.m./p.m.
lcl.datetime.date.format	string which includes 'D', 'd' and 'M' and two optional commas	Controls format of date string. D = day of week d = day M = month Up to two commas may be included. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: "D,,dM" is illegal.
lcl.datetime.date.longFormat	0,1	If 1, display the day and month in long format (Friday/November), otherwise, use abbreviations (Fri/Nov).
lcl.datetime.date.dateTop	0, 1	If 1, display date above time else display time above date.

User Preferences <up/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
up.headsetMode	0,1	0	If set to 1, the headset will be selected as the preferred transducer after its first use until the headset key is pressed again; otherwise, hands-free will be selected preferentially over the headset.
up.useDirectoryNames	0,1	0	If set to 1, the name fields of directory entries which match incoming calls will be used for caller identification display and in the call lists instead of the name provided through network signaling.
up.oneTouchVoiceMail	0, 1	0	If set to 1, the voice mail summary display is bypassed and voice mail is dialed directly (if configured).
up.welcomeSoundEnabled	0, 1	1	If set to 1, play welcome sound effect after a reboot.

Attribute	Permitted Values	Default	Interpretation
up.welcomeSoundOnWarmBootEnabled	0, 1	0	If set to 1, play welcome sound effect on warm as well as cold boots, otherwise only a cold boot will trigger the welcome sound effect.
up.localClockEnabled	0, 1	1	If set to 1, display the date and time on the idle display
up.backlight.onIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	Null	This parameter controls the intensity of the LCD backlight when it turns on during normal use of the phone. The default value is medium.
up.backlight.idleIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	Null	This parameter controls the intensity of the LCD backlight when the phone is idle. The default value is low. Note: If <i>idleIntensity</i> is set higher than <i>onIntensity</i> , it will be replaced with the <i>onIntensity</i> value.
up.idleTimeout	positive integer, seconds	Null	Timeout for the idle display or default call handling display. If set to 0, there is no timeout. If set to Null, the default timeout of 20 seconds is used. If set to value greater than 0, the timeout is for that number of seconds (maximum 65536).
up.mwiVisible	0 - Disabled 1 - Enabled	0	If set to 0 or Null, there is no MWI for registration x (SIP 2.1.0 and 2.1.1 behavior). If set to 1, <code>msg.mwi.x.callBackMode</code> is set to disabled. MWI notification will be displayed for registration x (Pre-SIP 2.1.0 behavior).
up.handsfreeMode	0 - Disabled 1 - Enabled	1	If set to 1 or Null, hands-free speakerphone is enabled. If set to 0, hands-free speakerphone is disabled.

Tones <tones/>

This attribute describes configuration items for the tone resources available in the phone.

This attribute includes:

- [Dual Tone Multi-Frequency <DTMF/>](#)
- [Chord-Sets <chord/>](#)

Dual Tone Multi-Frequency <DTMF/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tone.dtmf.level	-33 to -3	-15	Level of the high frequency component of the DTMF digit measured in dBm0; the low frequency tone will be two dB lower.
tone.dtmf.onTime	positive integer	50	When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the tones will be generated for; this is also the minimum time the tone will be played for when dialing manually (even if key press is shorter).
tone.dtmf.offTime	positive integer	50	When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the phone will pause between digits; this is also the minimum inter-digit time when dialing manually.
tone.dtmf.chassis.masking	0, 1	0	If set to 1, DTMF tones will be substituted with a non-DTMF pacifier tone when dialing in hands-free mode. This prevents DTMF digits being broadcast to other surrounding telephony devices or being inadvertently transmitted in-band due to local acoustic echo. Note: <i>tone.dtmf.chassis.masking should only be enabled when tone.dtmf.viaRtp is disabled.</i>
tone.dtmf.stim.pac.offHookOnly	0, 1	0	Not currently used.

Attribute	Permitted Values	Default	Interpretation
tone.dtmf.viaRtp	0, 1	1	If set to 1, encode DTMF in the active RTP stream, otherwise, DTMF may be encoded within the signaling protocol only when the protocol offers the option. Note: <i>tone.dtmf.chassis.masking should be enabled when tone.dtmf.viaRtp is disabled.</i>
tone.dtmf.rfc2833Control	0, 1	1	If set to 1, the phone will indicate a preference for encoding DTMF through RFC 2833 format in its Session Description Protocol (SDP) offers by showing support for the phone-event payload type; this does not affect SDP answers, these will always honor the DTMF format present in the offer since the phone has native support for RFC 2833.
tone.dtmf.rfc2833Payload	96-127	101	The phone-event payload encoding in the dynamic range to be used in SDP offers.

Chord-Sets <chord/>

Chord-sets are the building blocks of sound effects that use synthesized rather than sampled audio (most call progress and ringer sound effects). A chord-set is a multi-frequency note with an optional on/off cadence. A chord-set can contain up to four frequency components generated simultaneously, each with its own level.

There are three blocks of chord sets:

- callProg (used for call progress sound effect patterns)
- ringer
- misc (miscellaneous)

All three blocks use the same chord set specification format.

In the following table, *x* is the chord-set number and *cat* is one of callProg, ringer, or misc.

Attribute	Permitted Values	Interpretation
tone.chord.cat.x.freq.y	0-1600	Frequency for this component in Hertz; up to four chord-set components can be specified (y=1, 2, 3, 4).
tone.chord.cat.x.level.y	-57 to 3	Level of this component in dBm0.
tone.chord.cat.x.onDur	positive integer	On duration in milliseconds, 0=infinite.
tone.chord.cat.x.offDur	positive integer	Off duration in milliseconds, 0=infinite.
tone.chord.cat.x.repeat	positive integer	Specifies how many times the ON/OFF cadence is repeated, 0=infinite.

Sampled Audio for Sound Effects <saf/>

The following sampled audio WAVE file (.wav) formats are supported:

- mono 8 kHz G.711 μ -Law
- G.711 A-Law
- L16/16000 (16-bit, 16 kHz sampling rate, mono)

Note

L16/16000 is not supported on SoundPoint IP 301phones, and SoundStation IP 4000 phones.

The phone uses built-in wave files for some sound effects. The built-in wave files can be replaced with files downloaded from the boot server or from the Internet, however, these are stored in volatile memory so the files will need to remain accessible should the phone need to be rebooted. Files will be truncated to a maximum size of 300 kilobytes.

In the following table, *x* is the sampled audio file number.

Attribute	Permitted Values	Interpretation
saf.x	Null OR valid path name OR an RFC 1738-compliant URL to a HTTP, FTP, or TFTP wave file resource. Note: Refer to the above wave file format restrictions.	If Null, the phone will use a built-in file. If set to a path name, the phone will attempt to download this file at boot time from the boot server. If set to a URL, the phone will attempt to download this file at boot time from the Internet. Note: A TFTP URL is expected to be in the format: <i>ftp://<host>/[pathname]<filename></i> , for example: <i>ftp://somehost.example.com/sounds/example.wav</i> .

The following table defines the default usage of the sampled audio files with the phone:

Sampled Audio File	Default use within phone (pattern reference)
1	Welcome Sound Effect (se.pat.misc.7)
2	Ringer 13 (se.pat.ringer.13)
3	Ringer 14 (se.pat.ringer.14)
4	Ringer 15 (se.pat.ringer.15)
5	Ringer 16 (se.pat.ringer.16)
6	Ringer 17 (se.pat.ringer.17)
7	Ringer 18 (se.pat.ringer.18)
8	Ringer 19 (se.pat.ringer.19)
9	Ringer 20 (se.pat.ringer.20)
10	Ringer 21 (se.pat.ringer.21)
11	Ringer 22 (se.pat.ringer.22)
12-24	Not used.

Sound Effects <se/>

The phone uses both synthesized (based on the chord-sets, refer to [Chord-Sets <chord/>](#) on page A-26) and sampled audio sound effects. Sound effects are defined by patterns: rudimentary sequences of chord-sets, silence periods, and wave files.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
se.stutterOnVoiceMail	0, 1	1	If set to 1, stuttered dial tone is used in place of normal dial tone to indicate that one or more messages (voice mail) are waiting at the message center.
se.appLocalEnabled	0, 1	1	If set to 1, local user interface sound effects such as confirmation/error tones, will be enabled.

This attribute also includes:

- [Patterns <pat/>](#)
- [Ring type <rt/>](#)

Patterns <pat/>

Patterns use a simple script language that allows different chord sets or wave files to be strung together with periods of silence. The script language uses the following instructions:

Instruction	Meaning	Example
sampled (n)	Play sampled audio file n	se.pat.callProg.x.inst.y.type = "sampled" (sampled audio file instruction type) se.pat.callProg.x.inst.y.value = "3" (specifies sampled audio file 3)
chord (n, d)	Play chord set n (d is optional and allows the chord set ON duration to be overridden to d milliseconds)	se.pat.callProg.x.inst.y.type = "chord" (chord set instruction type) se.pat.callProg.x.inst.y.value = "3" (specifies call progress chord set 3) se.pat.callProg.x.inst.y.param = "2000" (override ON duration of chord set to 2000 milliseconds)
silence (d)	Play silence for d milliseconds (Rx audio is not muted)	se.pat.callProg.x.inst.y.type = "silence" (silence instruction type) se.pat.callProg.x.inst.y.value = "300" (specifies silence is to last 300 milliseconds)
branch (n)	Advance n instructions and execute that instruction (n must be negative and must not branch beyond the first instruction)	se.pat.callProg.x.inst.y.type = "branch" (branch instruction type) se.pat.callProg.x.inst.y.value = "-5" (step back 5 instructions and execute that instruction)

Note

Currently, patterns that use the *sampled* instruction are limited to the following format: *sampled* followed by optional *silence* and optional *branch* back to the beginning.

In the following table, *x* is the pattern number, *y* is the instruction number. Both *x* and *y* need to be sequential. There are three categories of sound effect patterns: *callProg* ([Call Progress Patterns](#)), *ringer* ([Ringer Patterns](#)) and *misc* ([Miscellaneous Patterns](#)).

Attribute	Permitted Values	Interpretation	
se.pat.callProg.x.name	UTF-8 encoded string	Used for identification purposes in the user interface (currently used for ringer patterns only); for patterns that use a sampled audio file which has been overridden by a downloaded replacement, the se.pat.ringer.x.name parameter will be overridden in the user interface by the file names of the wave file.	
se.pat.callProg.x.inst.y.type	sampled OR chord OR silence OR branch	As above.	
se.pat.callProg.x.inst.y.value	integer	Instruction type: sampled chord silence branch	Interpretation: sampled audio file number chord set number silence duration in ms number of instructions to advance
se.pat.callProg.x.inst.y.param	positive integer	If instruction type is chord, this optional parameter specifies the on duration to be used, overriding the on duration specified in the chord-set definition.	

Call Progress Patterns

The following table maps call progress patterns to their usage within the phone.

Call progress pattern number	Use within phone
1	dial tone
2	busy tone
3	ring back tone
4	reorder tone
5	stuttered dial tone
6	call waiting tone

Call progress pattern number	Use within phone
7	alternate call waiting tone (distinctive)
8	confirmation tone
9	howler tone (off-hook warning)
10	record warning
11	message waiting tone
12	alerting
13	intercom announcement tone
14	barge-in tone
15	secondary dial tone

Ringer Patterns

The following table maps ringer pattern numbers to their default descriptions.

Ringer pattern number	Default description
1	Silent Ring
2	Low Trill
3	Low Double Trill
4	Medium Trill
5	Medium Double Trill
6	High Trill
7	High Double Trill
8	Highest Trill
9	Highest Double Trill
10	Beeble
11	Triplet
12	Ringback-style
13	Sampled audio file 2
14	Sampled audio file 3
15	Sampled audio file 4
16	Sampled audio file 5
17	Sampled audio file 6

Ringer pattern number	Default description
18	Sampled audio file 7
19	Sampled audio file 8
20	Sampled audio file 9
21	Sampled audio file 10
22	Sampled audio file 11

Note

Silent Ring will only provide a visual indication of an incoming call, but no audio indication.

Sampled audio files 1-21 all use the same built-in file unless that file has been replaced with a downloaded file. For more information, refer to [Sampled Audio for Sound Effects <saf/>](#) on page [A-27](#).

Miscellaneous Patterns

The following table maps miscellaneous patterns to their usage within the phone.

Miscellaneous pattern number	Use within phone
1	new message waiting indication
2	new instant message
3	Not used
4	local hold notification
5	positive confirmation
6	negative confirmation
7	welcome (boot up)

Ring type <rt/>

Ring type is used to define a simple class of ring to be applied based on some credentials that are usually carried within the network protocol. The ring class includes attributes such as call-waiting and ringer index, if appropriate. The ring class can use one of four types of ring that are defined as follows:

ring	Play a specified ring pattern or call waiting indication.
visual	Provide only a visual indication (no audio indication) of incoming call (no ringer needs to be specified).
answer	Provide auto-answer on incoming call.
ring-answer	Provide auto answer on incoming call after a ring period.

Note

The auto-answer on incoming call is currently only applied if there is no other call in progress on the phone at the time.

In the following table, x is the ring class number. The x index needs to be sequential.

Attribute	Permitted Values	Interpretation
se.rt.enabled	0,1	Set to 1 to enable the ring type feature within the phone, 0 otherwise.
se.rt.modification.enabled	0,1	Set to 1 to allow user modification through local user interface of the pre-defined ring type enabled for modification.
se.rt.x.name	UTF-8 encoded string	Used for identification purposes in the user interface.
se.rt.x.type	ring OR visual OR answer OR ring-answer	As defined in table above.
se.rt.x.ringer	integer - only relevant if the type is set to 'ring' or 'ring-answer'	The ringer index to be used for this class of ring. The ringer index should match one of Ringer Patterns on page A-31.
se.rt.x.callWait	integer - only relevant if the type is set to 'ring' or 'ring-answer'	The call waiting index to be used for this class of ring. The call waiting index should match one defined in Call Progress Patterns on page A-30.
se.rt.x.timeout	positive integer - only relevant if the type is set to 'ring-answer'. Default value is 2000.	The duration of the ring in milliseconds before the call is auto answered. If this field is omitted or is left blank, a value of 2000 is used.
se.rt.x.mod	0,1	Set to 1 if the user interface should allow for modification by the user of the ringer index used for this ring class.

Note

Modification of `se.rt.modification.enabled` and `se.rt.x.name` parameters through the user interface will be implemented in a future release.

Voice Settings <voice/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.txPacketFilter	0, 1	Null	Flag to determine whether or not narrowband Tx high-pass filtering should be enabled. If set to 1, narrowband Tx high-pass filter is enabled. If set 0 or Null, no Tx filtering is performed.

This attribute includes:

- [Voice Coding Algorithms <codecs/>](#)
- [Volume Persistence <volume/>](#)
- [Gains <gain/>](#)
- [Acoustic Echo Cancellation <aec/>](#)
- [Acoustic Echo Suppression <aes/>](#)
- [Background Noise Suppression <ns/>](#)
- [Automatic Gain Control <agc/>](#)
- [Receive Equalization <rxEq/>](#)
- [Transmit Equalization <txEq/>](#)
- [Voice Activity Detection <vad/>](#)
- [Quality Monitoring <quality monitoring/>](#)

Voice Coding Algorithms <codecs/>

The following voice codecs are supported:

Algorithm	MIME Type	Label	Bit Rate	Sample Rate	Frame Size	Effective Audio Bandwidth
G.711 μ -law	PMCU	G711mu	64 Kbps	8 Ksps	10ms - 80ms	3.5 KHz
G.711a-law	PCMA	G711A	64 Kbps	8 Ksps	10ms - 80ms	3.5 KHz
G.722	G722/8000	G722	64 Kbps	16 Ksps	10ms - 80ms	7 KHz
G.722.1	G722/16000	G7221	16 Kbps, 24 Kbps, 32 Kbps	16 Ksps	20ms - 80ms	7 KHz
G.729AB	G729	G729AB	8 Kbps	8 Ksps	10ms - 80ms	3.5 KHz

These codecs include:

- [Codec Preferences <codecPref/>](#)
- [Codec Profiles <audioProfile/>](#)

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.G711Mu	Null, 1-3	1	Specifies the codec preferences for SoundPoint IP 320, 330, 430, 500, 501, 600 and 601 platforms. 1 = highest 3 = lowest Null = do not use Give each codec a unique priority, this will dictate the order used in SDP negotiations.
voice.codecPref.G711A		2	
voice.codecPref.G729AB		3	
voice.codecPref.IP_300.G711Mu	Null, 1-3	1	Specifies the codec preferences for SoundPoint IP 301 models. Interpretation as above.
voice.codecPref.IP_300.G711A		2	
voice.codecPref.IP_300.G729AB		3	
voice.codecPref.IP_650.G711Mu	Null, 1-4	2	Specifies the codec preferences for the SoundPoint IP 550, 560, and 650 platform. Interpretation as above.
voice.codecPref.IP_650.G711A		3	
voice.codecPref.IP_650.G729AB		4	
voice.codecPref.IP_650.G722		1	

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_4000.G711Mu	Null, 1-3	1	Specifies the codec preferences for the SoundStation IP 4000 platform. Interpretation as above.
voice.codecPref.IP_4000.G711A		2	
voice.codecPref.IP_4000.G729AB		Null	Not supported by default so that G.711Mu and G.711A local conferences can be supported.

Codec Profiles <audioProfile/>

The following profile attributes can be adjusted for each of the five supported codecs. In the table, *x*=G711Mu, G711A, G722, G7221, and G729AB.

Attribute	Permitted Values	Interpretation
voice.audioProfile.x.payloadSize	10, 20, 30, ...80	Preferred Tx payload size in milliseconds to be provided in SDP offers and used in the absence ofptime negotiations. This is also the range of supported Rx payload sizes.
voice.audioProfile.x.jitterBufferMin	20, 40, 50, 60, ... (multiple of 10)	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter. The IP4000 values are the same as the IP30x values.
voice.audioProfile.x.jitterBufferShrink	10, 20, 30, ... (multiple of 10)	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (1000 ms) to minimize the delay on known good networks. Use larger values to minimize packet loss on networks with large jitter (3000 ms).
voice.audioProfile.x.jitterBufferMax	> jitterBufferMin, multiple of 10, <=300 for IP 320, 330, 430, 501,550, 600, 601, and 650 <= 200 for IP 301	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.

Volume Persistence <volume/>

The user's selection of the receive volume during a call can be remembered between calls. This can be configured per termination (handset, headset and hands-free/chassis). In some countries regulations exist which dictate that receive volume should be reset to nominal at the start of each call on handset and headset.

Attribute	Permitted Values	Default	Interpretation
voice.volume.persist.handset	0, 1	0	If set to 1, the receive volume will be remembered between calls.
voice.volume.persist.headset	0, 1	0	
voice.volume.persist.handsfree	0, 1	1	If set to 0, the receive volume will be reset to nominal at the start of each call.

Gains <gain/>

The default gain settings have been carefully adjusted to comply with the TIA-810-A digital telephony standard.



Polycom recommends that you do not change these values.

Attribute	Default
voice.gain.rx.analog.handset	0
voice.gain.rx.analog.headset	0
voice.gain.rx.analog.chassis	0
voice.gain.rx.analog.chassis.IP_300	-6
voice.gain.rx.analog.chassis.IP_330	0
voice.gain.rx.analog.chassis.IP_430	0
voice.gain.rx.analog.chassis.IP_601	6
voice.gain.rx.analog.chassis.IP_650	0
voice.gain.rx.analog.ringer	0
voice.gain.rx.analog.ringer.IP_300	-6
voice.gain.rx.analog.ringer.IP_330	0
voice.gain.rx.analog.ringer.IP_430	0
voice.gain.rx.analog.ringer.IP_601	6

Attribute	Default
voice.gain.rx.analog.ringer.IP_650	0
voice.gain.rx.digital.handset	-15
voice.gain.rx.digital.headset	-21
voice.gain.rx.digital.chassis	0
voice.gain.rx.digital.chassis.IP_330	6
voice.gain.rx.digital.chassis.IP_430	6
voice.gain.rx.digital.chassis.IP_4000	0
voice.gain.rx.digital.chassis.IP_601	0
voice.gain.rx.digital.chassis.IP_650	6
voice.gain.rx.digital.ringer	-21
voice.gain.rx.digital.ringer.IP_330	-12
voice.gain.rx.digital.ringer.IP_430	-12
voice.gain.rx.digital.ringer.IP_4000	-21
voice.gain.rx.digital.ringer.IP_601	-21
voice.gain.rx.digital.ringer.IP_650	-12
voice.gain.rx.analog.handset.sidetone	-20
voice.gain.rx.analog.headset.sidetone	-24
voice.gain.tx.analog.handset	6
voice.gain.tx.analog.headset	3
voice.gain.tx.analog.chassis	3
voice.gain.tx.analog.chassis.IP_300	0
voice.gain.tx.analog.chassis.IP_330	36
voice.gain.tx.analog.chassis.IP_430	36
voice.gain.tx.analog.chassis.IP_601	0
voice.gain.tx.analog.chassis.IP_650	36
voice.gain.tx.digital.handset	0
voice.gain.tx.digital.handset.IP_330	10
voice.gain.tx.digital.handset.IP_430	10
voice.gain.tx.digital.handset.IP_650	6
voice.gain.tx.digital.headset	0

Attribute	Default
voice.gain.tx.digital.headset.IP_330	10
voice.gain.tx.digital.headset.IP_430	10
voice.gain.tx.digital.headset.IP_650	6
voice.gain.tx.digital.chassis	3
voice.gain.tx.digital.chassis.IP_330	12
voice.gain.tx.digital.chassis.IP_430	12
voice.gain.tx.digital.chassis.IP_4000	0
voice.gain.tx.digital.chassis.IP_601	6
voice.gain.tx.digital.chassis.IP_650	12
voice.gain.tx.analog.preamp.handset	14
voice.gain.tx.analog.preamp.headset	23
voice.gain.tx.analog.preamp.chassis	32
voice.gain.tx.analog.preamp.chassis.IP_601	32
voice.handset.rxag.adjust.IP_330	1
voice.handset.rxag.adjust.IP_430	1
voice.handset.rxag.adjust.IP_650	1
voice.handset.txag.adjust.IP_330	18
voice.handset.txag.adjust.IP_430	18
voice.handset.txag.adjust.IP_650	18
voice.handset.sidetone.adjust.IP_330	3
voice.handset.sidetone.adjust.IP_430	3
voice.handset.sidetone.adjust.IP_650	0
voice.headset.rxag.adjust.IP_330	4
voice.headset.rxag.adjust.IP_430	1
voice.headset.rxag.adjust.IP_650	1
voice.headset.txag.adjust.IP_330	21
voice.headset.txag.adjust.IP_430	39
voice.headset.txag.adjust.IP_650	21

Attribute	Default
voice.headset.sidetone.adjust.IP_330	-3
voice.headset.sidetone.adjust.IP_430	-3
voice.headset.sidetone.adjust.IP_650	-3

Acoustic Echo Cancellation <aec/>

These settings control the performance of the speakerphone acoustic echo canceller.



Polycom recommends that you do not change these values.

Attribute	Default
voice.aec.hs.enable	0
voice.aec.hs.lowFreqCutOff	100
voice.aec.hs.highFreqCutOff	7000
voice.aec.hs.erlTab_0_300	-24
voice.aec.hs.erlTab_300_600	-24
voice.aec.hs.erlTab_600_1500	-24
voice.aec.hs.erlTab_1500_3500	-24
voice.aec.hs.erlTab_3500_7000	-24
voice.aec.hd.enable	0
voice.aec.hd.lowFreqCutOff	100
voice.aec.hd.highFreqCutOff	7000
voice.aec.hd.erlTab_0_300	-24
voice.aec.hd.erlTab_300_600	-24
voice.aec.hd.erlTab_600_1500	-24
voice.aec.hd.erlTab_1500_3500	-24
voice.aec.hd.erlTab_3500_7000	-24
voice.aec.hf.enable	1
voice.aec.hf.lowFreqCutOff	100
voice.aec.hf.highFreqCutOff	7000

Attribute	Default
voice.aec.hf.erlTab_0_300	-6
voice.aec.hf.erlTab_300_600	-6
voice.aec.hf.erlTab_600_1500	-6
voice.aec.hf.erlTab_1500_3500	-6
voice.aec.hf.erlTab_3500_7000	-6

Acoustic Echo Suppression <aes/>

These settings control the performance of the speakerphone acoustic echo suppressor.



Polycom recommends that you do not change these values.

Attribute	Default
voice.aes.hs.enable	0
voice.aes.hs.duplexBalance	7
voice.aes.hd.enable	0
voice.aes.hd.duplexBalance	0
voice.aes.hf.enable	1
voice.aes.hf.duplexBalance.0	7
voice.aes.hf.duplexBalance.1	7
voice.aes.hf.duplexBalance.2	6
voice.aes.hf.duplexBalance.3	6
voice.aes.hf.duplexBalance.4	5
voice.aes.hf.duplexBalance.5	4
voice.aes.hf.duplexBalance.6	4
voice.aes.hf.duplexBalance.7	3
voice.aes.hf.duplexBalance.8	2
voice.aes.hf.duplexBalance.IP_4000.0	10
voice.aes.hf.duplexBalance.IP_4000.1	9
voice.aes.hf.duplexBalance.IP_4000.2	8

Attribute	Default
voice.aes.hf.duplexBalance.IP_4000.3	7
voice.aes.hf.duplexBalance.IP_4000.4	6
voice.aes.hf.duplexBalance.IP_4000.5	5
voice.aes.hf.duplexBalance.IP_4000.6	4
voice.aes.hf.duplexBalance.IP_4000.7	3
voice.aes.hf.duplexBalance.IP_4000.8	2

Background Noise Suppression <ns/>

These settings control the performance of the transmit background noise suppression feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.ns.hs.enable	0
voice.ns.hs.signalAttn	-6
voice.ns.hs.silenceAttn	-9
voice.ns.hd.enable	0
voice.ns.hd.signalAttn	0
voice.ns.hd.silenceAttn	0
voice.ns.hf.enable	1
voice.ns.hf.signalAttn	-6
voice.ns.hf.silenceAttn	-9
voice.ns.hf.IP_4000.enable	1
voice.ns.hf.IP_4000.signalAttn	-6
voice.ns.hf.IP_4000.silenceAttn	-9

Automatic Gain Control <agc/>

These settings control the performance of the transmit automatic gain control feature.

Note

Automatic Gain Control will be implemented in a future release.



Polycom recommends that you do not change these values.

Attribute	Default
voice.agc.hs.enable	0
voice.agc.hd.enable	0
voice.agc.hf.enable	0

Receive Equalization <rxEq/>

These settings control the performance of the receive equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.rxEq.hs.IP_330.preFilter.enable	1
voice.rxEq.hs.IP_430.preFilter.enable	1
voice.rxEq.hs.IP_500.preFilter.enable	1
voice.rxEq.hs.IP_600.preFilter.enable	1
voice.rxEq.hs.IP_601.preFilter.enable	1
voice.rxEq.hs.IP_650.preFilter.enable	1
voice.rxEq.hs.IP_330.postFilter.enable	0
voice.rxEq.hs.IP_430.postFilter.enable	0
voice.rxEq.hs.IP_500.postFilter.enable	0
voice.rxEq.hs.IP_600.postFilter.enable	0
voice.rxEq.hs.IP_601.postFilter.enable	0

Attribute	Default
voice.rxEq.hs.IP_650.postFilter.enable	0
voice.rxEq.hd.IP_330.preFilter.enable	0
voice.rxEq.hd.IP_430.preFilter.enable	0
voice.rxEq.hd.IP_500.preFilter.enable	0
voice.rxEq.hd.IP_600.preFilter.enable	0
voice.rxEq.hd.IP_601.preFilter.enable	0
voice.rxEq.hd.IP_650.preFilter.enable	1
voice.rxEq.hd.IP_330.postFilter.enable	0
voice.rxEq.hd.IP_430.postFilter.enable	0
voice.rxEq.hd.IP_500.postFilter.enable	0
voice.rxEq.hd.IP_600.postFilter.enable	0
voice.rxEq.hd.IP_601.postFilter.enable	0
voice.rxEq.hd.IP_650.postFilter.enable	0
voice.rxEq.hf.IP_330.preFilter.enable	1
voice.rxEq.hf.IP_430.preFilter.enable	1
voice.rxEq.hf.IP_500.preFilter.enable	1
voice.rxEq.hf.IP_600.preFilter.enable	1
voice.rxEq.hf.IP_601.preFilter.enable	1
voice.rxEq.hf.IP_650.preFilter.enable	1
voice.rxEq.hf.IP_4000.preFilter.enable	0
voice.rxEq.hf.IP_330.postFilter.enable	0
voice.rxEq.hf.IP_430.postFilter.enable	0
voice.rxEq.hf.IP_500.postFilter.enable	1
voice.rxEq.hf.IP_600.postFilter.enable	1
voice.rxEq.hf.IP_601.postFilter.enable	1
voice.rxEq.hf.IP_650.postFilter.enable	0
voice.rxEq.hf.IP_4000.postFilter.enable	0

Transmit Equalization <txEq/>

These settings control the performance of the hands-free transmit equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.txEq.hs.IP_330.preFilter.enable	0
voice.txEq.hs.IP_430.preFilter.enable	0
voice.txEq.hs.IP_500.preFilter.enable	0
voice.txEq.hs.IP_600.preFilter.enable	0
voice.txEq.hs.IP_601.preFilter.enable	0
voice.txEq.hs.IP_650.preFilter.enable	1
voice.txEq.hs.IP_330.postFilter.enable	1
voice.txEq.hs.IP_430.postFilter.enable	1
voice.txEq.hs.IP_500.postFilter.enable	1
voice.txEq.hs.IP_600.postFilter.enable	1
voice.txEq.hs.IP_601.postFilter.enable	1
voice.txEq.hs.IP_650.postFilter.enable	1
voice.txEq.hd.IP_330.preFilter.enable	0
voice.txEq.hd.IP_430.preFilter.enable	0
voice.txEq.hd.IP_500.preFilter.enable	0
voice.txEq.hd.IP_600.preFilter.enable	0
voice.txEq.hd.IP_601.preFilter.enable	0
voice.txEq.hd.IP_650.preFilter.enable	1
voice.txEq.hd.IP_330.postFilter.enable	0
voice.txEq.hd.IP_430.postFilter.enable	0
voice.txEq.hd.IP_500.postFilter.enable	0
voice.txEq.hd.IP_600.postFilter.enable	0
voice.txEq.hd.IP_601.postFilter.enable	0
voice.txEq.hd.IP_650.postFilter.enable	0

Attribute	Default
voice.txEq.hf.IP_330.preFilter.enable	0
voice.txEq.hf.IP_430.preFilter.enable	0
voice.txEq.hf.IP_500.preFilter.enable	0
voice.txEq.hf.IP_600.preFilter.enable	0
voice.txEq.hf.IP_601.preFilter.enable	0
voice.txEq.hf.IP_650.preFilter.enable	1
voice.txEq.hf.IP_4000.preFilter.enable	0
voice.txEq.hf.IP_330.postFilter.enable	1
voice.txEq.hf.IP_430.postFilter.enable	1
voice.txEq.hf.IP_500.postFilter.enable	1
voice.txEq.hf.IP_600.postFilter.enable	1
voice.txEq.hf.IP_601.postFilter.enable	1
voice.txEq.hf.IP_650.postFilter.enable	1
voice.txEq.hf.IP_4000.postFilter.enable	0

Voice Activity Detection <vad/>

These settings control the performance of the voice activity detection (silence suppression) feature.

Attribute	Permitted Values	Default	Interpretation
voice.vadEnable	0, 1	0	If set to 1, enable VAD.
voice.vadThresh	integer from 0 to 30	15	The threshold for determining what is active voice and what is background noise in dB. This does not apply to G.729AB codec operation which has its own built-in VAD function.

Attribute	Permitted Values	Default	Interpretation
voice.vad.signalAnnexB	0, 1	Null	<p>If set to 1 and <code>voice.vadEnable</code> is set to 1, Annex B is used. A new line can be added to SDP depending on the setting of this parameter and the <code>voice.vadEnable</code> parameter.</p> <ul style="list-style-type: none"> If <code>voice.vadEnable</code> is set to 1, add attribute line <code>a=fmtp:18 annexb="yes"</code> below <code>a=rtpmap...</code> attribute line (where '18' could be replaced by another payload). If <code>voice.vadEnable</code> is set to 0, add attribute line <code>a=fmtp:18 annexb="no"</code> below <code>a=rtpmap...</code> attribute line (where '18' could be replaced by another payload). <p>If set to 0 or Null, there is no change to SDP.</p>

Quality Monitoring <quality monitoring/>

This attribute includes:

- [Central Report Collector <collector/>](#)
- [Alert Reports <alert/>](#)
- [Server <server/>](#)
- [RTCP-XR <rtcpxr/>](#)

Central Report Collector <collector/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.enable.periodic	0, 1	0	Enables generation of periodic quality reports throughout a call.
voice.qualityMonitoring.collector.enable.session	0, 1	0	Enables generation of a quality report at the end of each call.

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.enable.triggeredPeriodic	0, 1, 2	0	<p>Controls the generation of periodic quality reports triggered by alert states.</p> <p>If set to 0, alert states do not cause periodic reports to be generated.</p> <p>If set to 1, periodic reports will be generated when an alert state is critical.</p> <p>If set to 2, periodic reports will be generated when an alert state is either warning or critical.</p> <p>Note: This parameter is ignored when <code>qualityMonitoring.collector.enable.periodic</code> is set 1, since periodic reports are sent throughout the duration of a call.</p>
voice.qualityMonitoring.collector.period	5 to 20	20	The time interval between successive periodic quality reports.

Alert Reports <alert/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.alert.moslq.threshold.warning	Null, 15 to 40	Null	<p>Threshold value of listening MOS score (MOS-LQ) that causes phone to send a warning alert quality report. Configure the desired MOS value multiplied by 10. If set to Null, warning alerts are not generated due to MOS-LQ.</p> <p>For example, a configured value of 35 corresponds to the MOS score 3.5.</p>
voice.qualityMonitoring.collector.alert.moslq.threshold.critical	Null, 15 to 40	Null	<p>Threshold value of listening MOS score (MOS-LQ) that causes phone to send a critical alert quality report. Configure the desired MOS value multiplied by 10. If set to Null, critical alerts are not generated due to MOS-LQ.</p> <p>For example, a configured value of 28 corresponds to the MOS score 2.8.</p>

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.alert.delay.threshold.warning	Null, 10 to 2000	Null	Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If set to Null, warning alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.
voice.qualityMonitoring.collector.alert.delay.threshold.critical	Null, 10 to 2000	Null	Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If set to Null, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.server.x.address	Dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1 as only one report collector is supported at this time.
voice.qualityMonitoring.collector.server.x.port	0, Null, 1 to 65535	5060	If port is 0 or Null, port 5060 will be used. Set x to 1 as only one report collector is supported at this time.

RTCP-XR <rtcpxr/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.enable.rtcpxr	0, 1	0	Enables generation of RTCP-XR packets.

Quality of Service <QOS/>

These settings control the Quality of Service (QOS) options.

This attribute includes:

- [Ethernet IEEE 802.1p/Q <ethernet/>](#)
- [IP TOS <IP/>](#)

Ethernet IEEE 802.1p/Q <ethernet/>

The following settings control the 802.1p/Q user_priority field:

- [RTP <RTP/>](#)
- [Call Control <callControl/>](#)
- [Other <other/>](#)

RTP <RTP/>

These parameters apply to RTP packets.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.rtp.user_priority	0-7	5	User-priority used for RTP packets.

Call Control <callControl/>

These parameters apply to call control packets, such as the network protocol signaling.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.callControl.user_priority	0-7	5	User-priority used for call control packets.

Other <other/>

These default parameter values are used for all packets which are not set explicitly.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.other.user_priority	0-7	2	User-priority used for packets that do not have a per-protocol setting.

IP TOS <ip/>

The following settings control the “type of service” field in outgoing packets:

- [RTP <rtp/>](#)
- [Call Control <callControl/>](#)

RTP <rtp/>

These parameters apply to RTP packets.

Attribute	Permitted Values	Default	Interpretation
qos.ip.rtp.dscp	0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Null	This parameter allows the DSCP of packets to be specified. If set to a value, this will override the other qos.ip.rtp... parameters. Default of Null which means the other qos.ip.rtp... parameters will be used.
qos.ip.rtp.min_delay	0, 1	1	If set to 1, set min-delay bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.max_throughput	0, 1	1	If set to 1, set max-throughput bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.max_reliability	0, 1	0	If set to 1, set max-reliability bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.min_cost	0, 1	0	If set to 1, set min-cost bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.precedence	0-7	5	If set to 1, set precedence bits in the IP TOS field of the IP header, or else don't set them.

Call Control <callControl/>

These parameters apply to call control packets, such as the network protocol signaling.

Attribute	Permitted Values	Default	Interpretation
qos.ip.callControl.dscp	0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Null	This parameter allows the DSCP of packets to be specified. If set to a value this will override the other qos.ip.callControl... parameters. Default of Null which means the other qos.ip.callControl... parameters will be used.
qos.ip.callControl.min_delay	0, 1	1	If set to 1, set min-delay bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.max_throughput	0, 1	0	If set to 1, set max-throughput bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.max_reliability	0, 1	0	If set to 1, set max-reliability bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.min_cost	0, 1	0	If set to 1, set min-cost bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.precedence	0-7	5	If set to 1, set precedence bits in the IP TOS field of the IP header, or else don't set them.

Basic TCP/IP <TCP_IP/>

This attribute includes:

- [Network Monitoring <netMon/>](#)
- [Time Synchronization <sntp/>](#)
- [Port <port/>](#)
- [Keep-Alive <keepalive/>](#)

Network Monitoring <netMon/>



Polycom recommends that you do not change these values.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default
tcpIpApp.netMon.enabled	0, 1	1
tcpIpApp.netMon.period	1 to 86400	30

Time Synchronization <sntp/>

The following table describes the parameters used to set up time synchronization and daylight savings time. The defaults shown will enable daylight savings time (DST) for North America.

Daylight savings defaults:

- Do not use fixed day, use first or last day of week in the month.
- Start DST on the second Sunday in March at 2 am.
- Stop DST on the first Sunday in November at 2 am.

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.resyncPeriod	positive integer	86400 (24 hours)	Time in seconds between Simple Network Time Protocol (SNTP) re-syncs.
tcpIpApp.sntp.address	valid host name or IP address	clock	Address of the SNTP server.

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.snmp.address.overrideDHCP	0, 1	0	These parameters determine whether configuration file parameters override DHCP parameters for the SNTP server address and Greenwich Mean Time (GMT) offset. If set to 0, DHCP values will override configuration file parameters. If set to 1, the configuration file parameters will override DHCP values.
tcpIpApp.snmp.gmtOffset	positive or negative integer	-28800 (Pacific time)	Offset in seconds of the local time zone from GMT. 3600 seconds = 1 hour
tcpIpApp.snmp.gmtOffset.overrideDHCP	0, 1	0	These parameters determine whether configuration file parameters override DHCP parameters for the SNTP server address and GMT offset. If set to 0, DHCP values will override configuration file parameters. If set to 1, the configuration file parameters will override DHCP values.
tcpIpApp.snmp.daylightSavings.enable	0, 1	1	If set to 1, apply daylight savings rules to displayed time.
tcpIpApp.snmp.daylightSavings.fixedDayEnable	0, 1	0	If set to 0, month, date, and dayOfWeek are used in DST date calculation. If set to 1, then only month and date are used.
tcpIpApp.snmp.daylightSavings.start.month	1-12	3 (March)	Month to start DST. Mapping: 1=Jan, 2=Feb, ..., 12=Dec

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.snmp.daylightSavings.start.date	1-31	8	If <code>fixedDayEnable</code> is set to 1, use as day of the month to start DST. If <code>fixedDayEnable</code> is set to 0, use the mapping: 1 = the first occurrence of a given day-of-the-week in a month, 8 = the second occurrence of a given day-of-the-week in a month, 15 = the third occurrence of a given day-of-the-week in a month, 22 = the fourth occurrence of a given day-of-the-week in a month
tcpIpApp.snmp.daylightSavings.start.time	0-23	2	Time of day to start DST in 24 hour clock. Mapping: 2=2 am, 14=2 pm
tcpIpApp.snmp.daylightSavings.start.dayOfWeek	1-7	1	Day of week to apply DST. Mapping: 1=Sun, 2=Mon, ..., 7=Sat
tcpIpApp.snmp.daylightSavings.start.dayOfWeek.lastInMonth	0, 1	0	If set to 1 and <code>fixedDayEnable</code> is set to 0, DST starts on the last day (specified by <code>start.dayOfWeek</code>) of the week in the month. The <code>start.date</code> is ignored.
tcpIpApp.snmp.daylightSavings.stop.month	1-12	11	Month to stop DST.
tcpIpApp.snmp.daylightSavings.stop.date	1-31	1	Day of the month to stop DST.
tcpIpApp.snmp.daylightSavings.stop.time	0-23	2	Time of day to stop DST in 24 hour clock.
tcpIpApp.snmp.daylightSavings.stop.dayOfWeek	1-7	1	Day of week to stop DST.
tcpIpApp.snmp.daylightSavings.stop.dayOfWeek.lastInMonth	0, 1	0	If set to 1 and <code>fixedDayEnable</code> set to 0, DST stops on the last day (specified by <code>stop.dayOfWeek</code>) of the week in the month. The <code>stop.date</code> is ignored.

Port <port/>

This attribute includes:

- RTP <rtp/>

RTP <rtp/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.port.rtp.filterByIp	0, 1	1	If set to 1, reject RTP packets arriving from (sent from) a non-negotiated (through SDP) IP address.
tcpIpApp.port.rtp.filterByPort	0, 1	0	If set to 1, reject RTP packets arriving from (sent from) a non-negotiated (through SDP) port.
tcpIpApp.port.rtp.forceSend	Null, 1024-65534	Null	When non-Null, send all RTP packets to, and expect all RTP packets to arrive on, the specified port. Note: both <i>tcpIpApp.port.rtp.filterByIp</i> and <i>tcpIpApp.port.rtp.filterByPort</i> must be enabled for this to work.
tcpIpApp.port.rtp.mediaPortRangeStart	Null, even integer from 1024-65534	Null	If set to Null, the value 2222 will be used for the first allocated RTP port, otherwise, the specified port will be used. Subsequent ports will be allocated from a pool starting with the specified port plus two up to a value of (start-port + 46), after which the port number will wrap back to the starting value.

Keep-Alive <keepalive/>

Allowing for the configuration of TCP keep-alive on SIP TLS connections, the phone can detect a failures quickly (in minutes) and attempt to re-register with the SIP call server (or its redundant pair).

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.keepalive.tcp.idleTransmitInterval	10 to 7200	Null	After idle x seconds, the keep-alive message is sent to the call server. If set to Null, the default value is 30 seconds. Note: If this parameter is set to a value that is out of range, the default value is used.
tcpIpApp.keepalive.tcp.noResponseTrasmitInterval	5 to 120	Null	If no response is received to keep-alive message, another keep-alive message is sent to the call server after x seconds. If set to Null, the default value to 20 seconds. Note: If this parameter is set to a value that is out of range, the default value is used.
tcpIpApp.keepalive.tcp.sip.tls.enable	0, 1	0	If set to 1, enable TCP keep-alive for SIP signalling connections that use TLS transport. If set to 0, disable TCP keep-alive for SIP signalling connections that use TLS transport.

Web Server <httpd/>

The phone contains a local web server for user and administrator features. This can be disabled for applications where it is not needed or where it poses a security threat. The web server supports both basic and digest authentication. The authentication user name and password are not configurable for this release.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
httpd.enabled	0, 1	1	If set to 1, the HTTP server will be enabled.

This attribute also includes:

[Configuration <cfg/>](#)

Configuration <cfg/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
httpd.cfg.enabled	0, 1	1	If set to 1, the HTTP server configuration interface will be enabled.
httpd.cfg.port	1-65535	80	Port is 80 for HTTP servers. Care should be taken when choosing an alternate port.

Call Handling Configuration <call/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.rejectBusyOnDnd	0, 1	1	If set to 1, reject all incoming calls with the reason "busy" if do-not-disturb is enabled. Note: This attribute is ignored when the line is configured as shared. The reason being that even though one party has turned on DND, the other person/people sharing that line do not necessarily want all calls to that number diverted away. Note: If server-based DND is enabled, this parameter is disabled.
call.enableOnNotRegistered	0, 1	1	If set to 1, calls will be allowed when the phone is not successfully registered, otherwise, calls will not be permitted without a valid registration.
call.offeringTimeOut	positive integer	60	Time in seconds to allow an incoming call to ring before dropping the call, 0=infinite. Note: The call diversion, no answer feature will take precedence over this feature if enabled. For more information, refer to No Answer <noanswer/> on page A-97.
call.ringBackTimeOut	positive integer	60	Time in seconds to allow an outgoing call to remain in the ringback state before dropping the call, 0=infinite.

Attribute	Permitted Values	Default	Interpretation
call.dialtoneTimeOut	Null, positive integer	60	Time in seconds to allow the dialtone to be played before dropping the call. If set to 0, the call is not dropped. If set to Null, call dropped after 60 seconds.
call.lastCallReturnString	string of maximum length 32	*69	The string sent to the server when the user selects the "last call return" action.
call.callsPerLineKey	1 to 24 OR 1 to 8	24 OR 8	For the SoundPoint IP 600, 601, and 650, the permitted range is 1 to 24 and the default is 24. For all other phones the permitted range is 1 to 8 and the default is 8. This is the number of calls that may be active or on hold per line key on the phone. Note that this may be overridden by the per-registration attribute of <code>reg.x.callsPerLineKey</code> . Refer to Registration <reg/> on page A-89.
call.stickyAutoLineSeize	Null, 0, 1	0	If set to 1, makes the phone use "sticky" line seize behavior. This will help with features that need a second call object to work with. The phone will attempt to initiate a new outgoing call on the same SIP line that is currently in focus on the LCD (this was the behavior in SIP 1.6.5). If set to 0 or Null, the feature is disabled (this was the behavior in SIP 1.6.6). Note: This may fail due to glare issues in which case the phone may select a different available line for the call.
call.stickyAutoLineSeize.onHook Dialing	Null, 0, 1	Null	If <code>call.stickyAutoLineSeize</code> is set to 1, this parameter has no effect. The regular <code>stickyAutoLineSeize</code> behavior is followed. If <code>call.stickyAutoLineSeize</code> is set to 0 or Null and this parameter is set to 1, this overrides the <code>stickyAutoLineSeize</code> behavior for hot dial only. (Any new call scenario seizes the next available line.) If <code>call.stickyAutoLineSeize</code> is set to 0 or Null and this parameter is set to 0 or Null, there is no difference between hot dial and new call scenarios. Note: A hot dial occurs on the line which is currently in the call appearance. Any new call scenario seizes the next available line.

Attribute	Permitted Values	Default	Interpretation
call.singleKeyPressConference	0,1	0	If set to 1, the conference will be setup after a user presses the Conference soft key or Conference key the first time. Also, all sound effects (dial tone, DTMF tone while dialing and ringing back) are heard by all existing participants in the conference. If set to 0 or Null, sound effects are only heard by conference initiator (old behavior). Only supported for SoundPoint IP 550, 560,650 . For all others, set to 0.
call.localConferenceCallHold	0, 1	0	If set to 0, a hold will happen for all legs when conference is put on hold. (old behavior). If set to 1, only the host is out of the conference, all other parties in conference continue to talk. (new behavior). If set to Null, the default value is 0. Only supported for SoundPoint IP 550, 560,650 . For all others, set to 0.
call.transfer.blindPreferred	0,1	Null	If set to 1, the Blind soft key appears as a transfer type. If set to 0 or Null, the Normal soft key appears. Note: This parameter is supported on the SoundPoint IP 330/320 only.

This attribute also includes:

- [Shared Calls <shared/>](#)
- [Hold, Local Reminder <hold/><localReminder/>](#)

[Shared Calls <shared/>](#)

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.shared.disableDivert	0, 1	1	If set to 1, disable diversion feature for shared lines. Note: This feature is disabled on most call servers.
call.shared.seizeFailReorder	0, 1	1	If set to 1, play re-order tone locally on shared line seize failure.

Attribute	Permitted Values	Default	Interpretation
call.shared.oneTouchResume	0, 1	0	<p>If set to 1, when a shared line has a call on hold the remote user can press that line and resume the call. If more than one call is on hold on the line then the first one will be selected and resumed automatically.</p> <p>If set to 0, pressing the shared line will bring up a list of the calls on that line and the user can select which call the next action should be applied to.</p> <p>Note: This parameter affects the SoundStation IP 4000 phone. For other phones a quick press and release of the line key will resume a call whereas pressing and holding down the line key will show a list of calls on that line.</p>
call.shared.exposeAutoHolds	0, 1	0	<p>If set to 1, on a shared line, when setting up a conference, a re-INVITE will be sent to the server.</p> <p>If set to 0, no re-INVITE will be sent to the server.</p>

Hold, Local Reminder <hold/><localReminder/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.hold.localReminder.enabled	0, 1	0	If set to 1, periodically notify the local user that calls have been on hold for an extended period of time.
call.hold.localReminder.period	non-negative integer	60	Time in seconds between subsequent reminders.
call.hold.localReminder.startDelay	non-negative integer	90	Time in seconds to wait before the initial reminder.

Directory <dir/>

This attribute includes:

- [Local Directory <local/>](#)
- [Corporate Directory <corp/>](#)

Local Directory <local/>

The local directory is stored in either flash memory or RAM on the phone. The local directory size is limited based on the amount of flash memory in the phone. (Different phone models have variable flash memory.)

When the volatile storage option is enabled, ensure that a properly configured boot server that allows uploads is available to store a back-up copy of the directory or its contents will be lost when the phone reboots or loses power.

Attribute	Permitted Values	Default	Interpretation
dir.local.volatile.2meg	0, 1	0	Attribute applies to platforms with 2 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.maxSize.2meg	1 to 20	20	Attribute applies to platforms with 2 Mbytes of flash memory. Maximum size in Kbytes of non-volatile storage that the directory will be permitted to consume.
dir.local.volatile.4meg	0, 1	0	Applies to platforms with 4 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.maxSize.4meg	1 to 50	50	Applies to platforms with 4 Mbytes of flash memory. Maximum size in Kbytes of non-volatile storage that the directory will be permitted to consume.
dir.local.volatile.maxSize	1 to 100	100	Maximum size in Kbytes of volatile storage that the directory will be permitted to consume. Note: For the SoundPoint IP 650 platform, this value is internally replaced by 2X the value.
dir.local.volatile.8meg	0, 1	0	Attribute applies only to platforms with 8 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.

Attribute	Permitted Values	Default	Interpretation
dir.local.nonVolatile.maxSize.8meg	1 to 100	100	Attribute applies only to platforms with 8 Mbytes of flash memory. This is the maximum size of non-volatile storage that the directory will be permitted to consume.
dir.local.readonly	0, 1	1	Specifies whether or not local contact directory is read only. If set to 0 or Null, the local contact directory is editable. If set to 1, the local contact directory is read only.
dir.search.field	0, 1	Null	Specifies how to search the contact directory. If set to 1, search by contact's first name. If set to 0, search by contact's last name.

Corporate Directory <corp/>

A portion of the corporate directory is stored in flash memory on the phone. The size is based on the amount of flash memory in the phone. (Different phone models have variable flash memory.)

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dir.corp.address	dotted-decimal IP address or host name	Null	The IP address or host name of the LDAP server interface to the corporate directory.
dir.corp.port	0, Null, 1 to 65535	389 (TCP) 636 (TLS)	This parameter is used to specify the port to connect to on the server, if a full URL is not provided.
dir.corp.transport	TCP, TLS, Null	TCP	This parameter is used to specify whether a TCP or TLS connection is made with the server, if a full URL is not provided.
dir.corp.baseDN	UTF-8 encoded string	Null	The base domain name is the starting point for making queries on the LDAP server.
dir.corp.user	UTF-8 encoded string	Null	The username used to authenticate to the LDAP server.

Attribute	Permitted Values	Default	Interpretation
dir.corp.password	UTF-8 encoded string	Null	The password used to authenticate to the LDAP server.
dir.corp.filterPrefix	UTF-8 encoded string	Null	Predefined filter string. If set to Null or invalid, "(objectclass=person)" is used.
dir.corp.attribute.x.name	UTF-8 encoded string	Null	The name of the attribute to match on the server. Each name must be unique, however, an LDAP entry can have multiple attributes with the same name. Up to eight attributes can be configured (x = 1 to 8).
dir.corp.attribute.x.label	UTF-8 encoded string	Null	A UTF-8 encoded string that is used as the label when data is displayed.
dir.corp.attribute.x.type	first_name, last_name, phone_number, SIP_address, URL, other	Null	This parameter defines how the attribute is interpreted by the phone. Entries can have multiple attributes of the same type. Type 'other' is used for display purposes only. If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the corporate directory.
dir.corp.attribute.x.sticky	0, 1	0	If set to 0 or Null, the filter criteria for this attribute is reset after a reboot. If set to 1, the filter criteria for this attribute is retained through a reboot. Such attributes are denoted with a "*" before the label when displayed on the phone.
dir.corp.attribute.x.filter	UTF-8 encoded string	Null	The filter string for this attribute, which is edited when searching.
dir.corp.backGroundSync	0, 1	1	If set to 0, there will be no background downloading from the LDAP server. If set to 1 or Null, there will be background downloading of data from the LDAP server.

Attribute	Permitted Values	Default	Interpretation
dir.corp.backGroundSync.period	60 to 65535, seconds	300	The corporate directory cache is refreshed after the corporate directory feature has not been used for this period of time.
dir.corp.viewPersistence	0, 1	0	If set to 0 or Null, the browse position in the data on the LDAP server and the attribute filters are reset for subsequent usage of the corporate directory. If set to 1, the browse position in the data and the attribute filters are retained for subsequent usage of the corporate directory.
dir.corp.leg.viewPersistence	0, 1	1	This parameter is the same as dir.corp.viewPersistence. Note: For SoundPoint IP 301, 501, 600, and 601 legacy phones, use "leg" tagged parameter. This prevents slow behaviour after exiting from the corporate directory.

Presence <pres/>

The parameter `pres.reg` is the line number used to send SUBSCRIBE. If this parameter is missing, the phone will use the primary line to send SUBSCRIBE.

Attribute	Permitted Values	Default	Interpretation
pres.reg	positive integer	1	Specifies the line/registration number used to send SUBSCRIBE for presence. Must be a valid line/registration number. If the number is not a valid line/registration number, it is ignored.
pres.idleSoftkeys	0, 1	Null	If set to Null or 0, the presence idle soft keys (MyStat and Buddies) do not appear. If set to Null or 1, the presence idle soft keys appear.

Fonts

Note

This section does not apply to the SoundPoint IP 301 phones.

These settings control the phone's ability to dynamically load an external font file during boot up. Loaded fonts can either overwrite pre-existing fonts embedded within the software (not recommended) or can extend the phone's font support for Unicode ranges not already embedded. The font file must be a Microsoft **.fnt** or **.fon** file format. (**.fon** file format is a collection of **.fnt** fonts grouped together within a single file.) The font file name must follow a specific pattern as described:

- Font filename:
 <fontName>_<fontHeightInPixels>_<fontRange>.<fontExtension>
- <fontName> is a free string of characters that typically carries the meaning of the font. Examples are "fontFixedSize" for a fixed-size font, or "fontProportionalSize" for a proportional size font.
- <fontHeightInPixels> describes the font height in number of screen pixels.
- <fontRange> describes the Unicode range covered by this font. Since **.fnt** or **.fon** are 256 characters based blocks, the <fontRange> is Uxx00_UxFF (**.fnt** file) or Uxx00_UyyFF (**.fon** file). For more information, refer to [Multilingual User Interface](#) on page 4-29.
- <fontExtension> describes the file type. Either **.fnt** for single 256 characters font or **.fon** for multiple **.fnt** files.

If it is necessary to overwrite an existing font, use these <fontName>_<fontHeightInPixels>:

SoundPoint IP 320, 330, 430, 500 and 501

"fontProp_10" This is the font used widely in the current implementation.

"fontPropSoftkey_10" This is the soft key specific font.

SoundPoint IP 550, 600, 601, and 650

"fontProp_19" This is the font used widely in the current implementation including for soft keys.

"fontProp_26" This is the font used to display time (but not date).

"fontProp_x" This is a small font used for the CPU/Load/Net utilization graphs, this is the same as the "fontProp_10" for the SoundPoint IP 500.

If the <fontName>_<fontHeightInPixels> does not match any of the names above, then the downloaded font will be applied against all fonts defined in the phone, which means that you may lose the benefit of fonts being calibrated differently depending on their usage. For example, the font used to display the

time on the SoundPoint IP 600 is a large font, larger than the one used to display the date, and if you overwrite this default font with a unique font, you lose this size aspect. For example:

- to overwrite the font used for SoundPoint IP 500 soft keys for ASCII, the name should be **fontPropSoftkey_10_U0000_U00FF.fnt** .
- to add support for a new font that will be used everywhere and that is not currently supported. For example, for the Eastern/Central European Czech language, this is Unicode range 100-17F, the name could be **fontCzechIP500_10_U0100_U01FF.fnt** and **fontCzechIP600_19_U0100_U01FF.fnt** .

When defining a single **.fon** file, there is a need for a *font delimiter*, currently “Copyright Polycom Canada Ltd” is used as an embedded delimiter, but this can be configured using **font.delimiter**. The font delimiter is important to retrieve the different scrambled **.fnt** blocks. This font delimiter must be placed in the “copyright” attribute of the **.fnt** header. **.fon** files are useful if you want to include support for a large number of font ranges at once; otherwise, if simply adding or changing a few fonts currently in use, multiple **.fnt** files are recommended since they are easier to work with individually.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.delimiter	string up to 256 ASCII characters	Null	Delimiter required to retrieve different grouped .fnt blocks.

This attribute also includes:

- [IP_330 font <IP_330/>](#)
- [IP_400 font <IP_400/>](#)
- [IP_500 font <IP_500/>](#)
- [IP_600 font <IP_600/>](#)

[IP_330 font <IP_330/>](#)

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_330.x.name	fontName_height_Uxx00_UyyFF.fon OR fontName_height_Uxx00_UxxFF.fnt	Null	Defines the font file that will be loaded from boot server during boot up. Note: When several <i>font.IP_330.x.name</i> are defined, the index <i>x</i> must follow consecutive increasing order.

IP_400 font <IP_400/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_400.x.name	fontName_height_Uxx00_UyyFF.fon OR fontName_height_Uxx00_UxxFF.fnt	Null	Defines the font file that will be loaded from boot server during boot up. Note: When several font.IP_430.x.name are defined, the index x must follow consecutive increasing order.

IP_500 font <IP_500/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_500.x.name	fontName_height_Uxx00_UyyFF.fon OR fontName_height_Uxx00_UxxFF.fnt	Null	Defines the font file that will be loaded from boot server during boot up. Note: When several font.IP_500.x.name are defined, the index x must follow consecutive increasing order.

IP_600 font <IP_600/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_600.x.name	fontName_height_Uxx00_UyyFF.fon OR fontName_height_Uxx00_UxxFF.fnt	Null	Defines the font file that will be loaded from boot server during boot up. Note: When several font.IP_600.x.name are defined, the index x must follow consecutive increasing order.

Keys <key/>

These settings control the scrolling behavior of keys and can be used to change key functions.

Attribute	Permitted Values	Default	Interpretation
key.scrolling.timeout	positive integer	1	The time-out after which a key that is enabled for scrolling will go into scrolling mode until the key is released. Keys enabled for scrolling are menu navigation keys (left, right, up, down arrows), volume keys, and some context-specific soft keys. The value is an integer multiple of 500 milliseconds (1=500ms).

SoundPoint IP 301, 320, 330, 430, 501, 550, 560, 600, 601, and 650 and SoundStation IP 4000 key functions can be changed from the factory defaults, although this is typically not necessary. For each key whose function you wish to change, add an XML attribute in the format described in the following table to the <keys .../> element of the configuration file. These will override the built-in assignments.



Polycom does not recommend the remapping for keys.

In the following table, x =IP_300, IP_330, IP_430, IP_500, IP_550, IP_600, IP_650, and IP_4000 and y is the key number. Note that IP_300 parameters affect SoundPoint IP 301 phones, IP_330 parameters affect SoundPoint IP 320 and 330 phones, IP_430 parameters affect SoundPoint IP 430 phones, IP_500 parameters affect SoundPoint IP 501 phones, IP_550 parameters affect SoundPoint IP 550 and 560 phones, IP_600 parameters affect SoundPoint IP 600 and 601 phones, IP_650 parameters affect SoundPoint IP 650 phones, IP_4000 parameters affect the SoundStation IP 4000 phones. IP 300: $y=1-35$; IP 330: $y=1-34$; IP 430: $y=1-35$; IP 500: $y=1-40$; IP 550: $y=1-40$; IP 600: $y=1-42$; IP_650: $y=1-42$; IP_4000: $y=1-29$.

Attribute	Permitted Values	Interpretation
key.x.y.function.prim	Functions listed below.	Sets the function for key y on platform x .
key.x.y.subPoint.prim	positive integer	Sets the sub-identifier for key functions with a secondary array identifier such as SpeedDial.

The following table lists the functions that are available:

Functions			
ArrowDown	Dialpad5	Line2	Select
ArrowLeft	Dialpad6	Line3	Setup
ArrowRight	Dialpad7	Line4	SoftKey1
ArrowUp	Dialpad8	Line5	SoftKey2
BuddyStatus	Dialpad9	Line6	SoftKey3
CallList	DialpadStar	Messages	SoftKey4
Conference	DialpadPound	Menu	SpeedDial
Delete	Directories	MicMute	SpeedDialMenu
Dialpad0	DoNotDisturb	MyStatus	Transfer
Dialpad1	Handsfree	Null	VolDown
Dialpad2	Headset	Offline	VolUp
Dialpad3	Hold	Redial	
Dialpad4	Line1	Release	

Backgrounds <bg/>

The backgrounds used by the SoundPoint IP 550, 560, and 650 phones are defined in this section. In the following table, x=1 to 4.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
bg.hiRes.gray.pat.solid.x.name	any string	White	Solid pattern name. For x=1: White, x=2: Light Gray, x=3, 4: Null
bg.hiRes.gray.pat.solid.x.red	0 to 255		The screen background layouts. For x=1, red (255), green, (255), blue (255) For x=2, red (160), green (160), blue (160) For x=3 and 4, all values are Null.
bg.hiRes.gray.pat.solid.x.green			
bg.hiRes.gray.pat.solid.x.blue			
bg.hiRes.gray.bm.x.name	any string		User defined graphic files. For x=1, string is "Leaf.bmp" For x=2, string is "World.bmp"

Bitmaps <bitmap/>

The bitmaps used by each phone model are defined in this section.

Platform <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/>

In the following table, $x=IP_300, IP_330, IP_400, IP_500, IP_600, IP_4000$ and y is the bitmap number. Note that IP_300 parameters affect SoundPoint IP 301 phones, IP_330 parameters affect SoundPoint IP 320 and 330 phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_500 parameters affect SoundPoint IP 501 phones, IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, and 650 phones, IP_4000 parameters affect SoundStation IP 4000 phones.

Attribute	Permitted Values	Interpretation
bitmap.x.y.name	The name of a bitmap to be used.	This is the name of a bitmap to be used for creating an animation. If the bitmap is to be downloaded from the boot server, its name must: <ul style="list-style-type: none"> • Be different from any name already in use in sip.cfg. • Match the name of the corresponding <fileName>.bmp to be retrieved from the boot server.

Indicators <ind/>

The following indicators are used by the phone:

- Animations <anim/> <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/>
- Patterns <pattern/>
- Classes <class/>
- Assignments

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
ind.idleDisplay.mode	1 (default), 2, 3	Null	The idle display animation screen layouts. For example, for the SoundPoint IP 330/320: <ul style="list-style-type: none"> If set to 1 or Null, the idle display animation size is 87 x 11 pixels. If set to 2, the idle display animation size is 87 x 22 pixels. If set to 3, the idle display animation size is 102 x 22 pixels.
ind.idleDisplay.enabled	0, 1	0	If set to 1, the idle display may support presentation of a custom animation if configured in the animation section of sip.cfg .

Animations <anim/> <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/>

This section defines bitmap animations composed of bitmap/duration couples. In the following table, $x=IP_300, IP_330, IP_400, IP_500, IP_600, IP_4000$, y is the animation number, z is the step in the animation. Note that IP_300 parameters affect SoundPoint IP 301 phones, IP_330 parameters affect SoundPoint IP 320 and 330 phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_500 parameters affect SoundPoint IP 501 phones and IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, and 650 phones, IP_4000 parameters affect SoundStation IP 4000 phones.

Note

As of SIP 2.2.0, a maximum of 24 frames per animation is supported.

Attribute	Permitted Values	Interpretation
ind.anim.x.y.frame.z.bitmap	A bitmap name defined previously.	Bitmap to use. Note that it must be defined already, refer to Platform <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/> on page A-71.
ind.anim.x.y.frame.z.duration	positive integer	Duration in milliseconds for this step. 0=infinite.

Patterns <pattern/>

This section defines patterns for the LED indicators. In the following table, *x* is the pattern number, *y* is the step in the pattern.

Attribute	Permitted Values	Interpretation
ind.pattern.x.step.y.state	On or Off	Turn LED on or off for this step.
ind.pattern.x.step.y.duration	positive integer	Duration in milliseconds for this step. 0=infinite
ind.pattern.x.step.y.colour	Red or Green (default is Red if not specified)	For bi-color LEDs, specify color.

Classes <class/>

This section defines the available classes for the LED and graphical icon indicator types. In the following table, *x* is the class number, *y* is the identifier of the state number for that class.

Attribute	Permitted Values	Interpretation
ind.class.x.state.y.index	positive integer	For LED type indicators, index refers to the pattern index, such as index <i>x</i> in the Patterns <pattern/> tag above. For Graphic Icon type indicators, index refers to the animation index, such as index <i>y</i> in the Animations <anim/> <IP_300/> , <IP_330/> , <IP_400/> , <IP_500/> , <IP_600/> , <IP_4000/> tag above.

Assignments

This attribute assigns a type and a class to an indicator. In the case of the Graphic Icon type, it also assigns a physical location and size in pixels on the LCD display (refer to the following section). In the case of the LED type, it assigns a physical LED number (refer to [Graphic Icons <gi/>](#) [<IP_300/>](#), [<IP_330>](#), [<IP_400/>](#), [<IP_500/>](#), [<IP_600/>](#), [<IP_4000/>](#) on page A-74).

LEDs <led/>

In the following table, *x* is the LED number.

Attribute	Permitted Values	Interpretation
ind.led.x.index		This is for internal usage only and should not be changed (this is the logical index).
ind.led.x.class	positive integer	Assigns the class (defined in Classes <class/> on page A-73) for this indicator.
ind.led.x.physNum		This maps the logical index to a specific physical LED.

Graphic Icons <gi/> <IP_300/>, <IP_330/>, <IP_400/>, <IP_500/>, <IP_600/>, <IP_4000/>

In the following table, *x*=IP_300, IP_330, IP_400, IP_500, IP_600, IP_4000, *y* is the graphic icon number. Note that IP_300 parameters affect SoundPoint IP 301 phones, IP_330 parameters affect SoundPoint IP 320 and 330 phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_500 parameters affect SoundPoint IP 501 phones, and IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, and 650 phones, IP_4000 parameters affect SoundStation IP 4000 phones.

Attribute	Permitted Values	Interpretation
ind.gi.x.y.index		This is for internal usage only and should not be changed (this is the logical index).
ind.gi.x.y.class	positive integer	Assigns the class (defined in Classes <class/> on page A-73) for this indicator.
ind.gi.x.y.physX	IP 300: 0-19 IP 330: 0-101 IP 400: 0-122 IP 500: 0-159 IP 600: 0-319 IP 4000: 0-247	For Graphic Icon type indicators, this is the x-axis location of the upper left corner of the indicator measured in pixels from left to right.

Attribute	Permitted Values	Interpretation
ind.gi.x.y.physY	IP 300: 0-3 IP 330: 0-19 IP 400: 0-45 IP 500: 0-79 IP 600: 0-159 IP 4000: 0-67	For Graphic Icon type indicators, this is the y-axis location of the upper left corner of the indicator measured in pixels from top to bottom.
ind.gi.x.y.physW	IP 300: n/a IP 330: 1-87 IP 400: 1-102 IP 500: 1-160 IP 600: 1-320 IP 4000: 1-248	For Graphic Icon type indicators, this is the width of the indicator measured in pixels.
ind.gi.x.y.physH	IP 300: n/a IP 330: 1-20 IP 400: 1-23 IP 500: 1-80 IP 600: 1-160 IP 4000: 1-68	For Graphic Icon type indicators, this is the height of the indicator measured in pixels.

Event Logging <log/>

Caution

Logging parameter changes can impair system operation. Do not change any logging parameters without prior consultation with Polycom Technical Support.

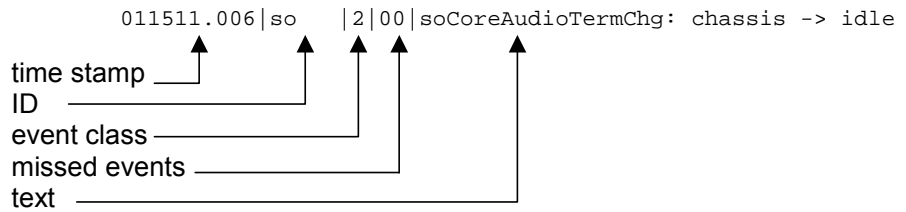
The event logging system supports the following classes of events:

Level	Interpretation
0	Debug only
1	High detail event class
2	Moderate detail event class
3	Low detail event class
4	Minor error - graceful recovery
5	Major error - will eventually incapacitate the system
6	Fatal error

Each event in the log contains the following fields separated by the | character:

- time or time/date stamp
- 1-5 character component identifier (such as "so")
- event class
- cumulative log events missed due to excessive CPU load
- free form text - the event description

Example:



Three formats are available for the event timestamp:

Type	Example
0 - seconds.milliseconds	011511.006 -- 1 hour, 15 minutes, 11.006 seconds since booting.
1 - absolute time with minute resolution	0210281716 -- 2002 October 28, 17:16
2 - absolute time with seconds resolution	1028171642 -- October 28, 17:16:42

Two types of logging are supported:

- [Basic Logging <level/><change/> and <render/>](#)
- [Scheduled Logging Parameters <sched/>](#)

Basic Logging <level/><change/> and <render/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
log.level.change.xxx	0-5	4	Control the logging detail level for individual components. These are the input filters into the internal memory-based log system. Possible values for xxx are so, app1, sip, sspsc, ssps, pps, net, cfg, cdp, pmt, ftp, ares, dns, cxss, httpd, rdisk, copy, slog, res, key, log, curl, rtos, mb, ib, sotet, ttrs, srtp, usb, efk, clink, ldap, and peer.
log.render.level	0-6	1	Specifies the lowest class of event that will be rendered to the log files. This is the output filter from the internal memory-based log system. The <code>log.render.level</code> maps to syslog severity as follows: 0 -> SeverityDebug (7) 1 -> SeverityDebug (7) 2 -> SeverityInformational (6) 3 -> SeverityInformational (6) 4 -> SeverityError (3) 5 -> SeverityCritical (2) 6 -> SeverityEmergency (0) 7 -> SeverityNotice (5) For more information, refer to Syslog Menu on page 3-12.
log.render.type	0-2	2	Refer to above table for timestamp type.
log.render.realtime	0, 1	1	Set to 1. Note: Polycom recommends that you do not change this value.
log.render.stdout	0, 1	1	Set to 1. Note: Polycom recommends that you do not change this value.
log.render.file	0, 1	1	Set to 1. Note: Polycom recommends that you do not change this value.

Attribute	Permitted Values	Default	Interpretation
log.render.file.size	positive integer, 1 to 179.5	16	Maximum local application log file size in Kbytes. When this size is exceeded, the file is uploaded to the boot server and the local copy is erased.
log.render.file.upload.period	positive integer	172800	Time in seconds between log file uploads to the boot server. Note: <i>The log file will not be uploaded if no new events have been logged since the last upload.</i>
log.render.file.upload.append	0, 1	1	If set to 1, use append mode when uploading log files to server. Note: <i>HTTP and TFTP don't support append mode unless the server is set up for this.</i>
log.render.file.upload.append.sizeLimit	positive integer	512	Maximum log file size on boot server in Kbytes.
log.render.file.upload.append.limitMode	delete, stop	delete	Behavior when server log file has reached its limit. delete=delete file and start over stop=stop appending to file

Scheduled Logging Parameters <sched/>

The phone can be configured to schedule certain advanced logging tasks on a periodic basis. These attributes should be set in consultation with Polycom Technical Support. Each scheduled log task is controlled by a unique attribute set starting with log.sched.x where x identifies the task.

Attribute	Permitted Values	Interpretation
log.sched.x.name	alphanumeric string	Name of an internal system command to be periodically executed. To be supplied by Polycom.
log.sched.x.level	0-5	Event class to assign to the log events generated by this command. This needs to be the same or higher than log.level.change.slog for these events to appear in the log.
log.sched.x.period	positive integer	Seconds between each command execution. 0=run once

Attribute	Permitted Values	Interpretation
log.sched.x.startMode	abs, rel	Start at <i>absolute</i> time or <i>relative</i> to boot.
log.sched.x.startTime	positive integer OR hh:mm	Seconds since boot when startMode is <i>rel</i> or the start time in 24-hour clock format when startMode is <i>abs</i> .
log.sched.x.startDay	1-7	When startMode is <i>abs</i> , specifies the day of the week to start command execution. 1=Sun, 2=Mon, ..., 7=Sat

Security <sec/>

This attribute's settings affect security aspects of the phone.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.tagSerialNo	0, 1	Null	If set to 1, the phone may advertise its serial number (Ethernet address) through protocol signaling. If set to 0 or Null, the phones does advertise its serial number.

This attribute also includes:

- [Encryption <encryption/>](#)
- [Password Lengths <pwd/><length/>](#)

Encryption <encryption/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.encryption.upload.dir	0, 1	0	<p>If set to 0, the phone-specific contact directory is uploaded to the server unencrypted regardless of how it was downloaded. This will replace whatever phone-specific contact directory is on the server even if it is encrypted.</p> <p>If set to 1, the phone-specific contact directory is uploaded encrypted regardless of how it was downloaded. This will replace whatever phone-specific contact directory is on the server even if it is unencrypted.</p>
sec.encryption.upload.override	0, 1	0	<p>If set to 0, the phone-specific configuration override file (<Ethernet Address>-phone.cfg) is uploaded unencrypted regardless of how it was downloaded. This will replace the override file on the server even if it is encrypted.</p> <p>If set to 1, the phone-specific configuration override file is uploaded encrypted regardless of how it was downloaded. This will replace the override file on the server even if it is unencrypted.</p>

Password Lengths <pwd/><length/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.pwd.length.admin	0-32	1	Password changes will need to be at least this long. Use 0 to allow null passwords.
sec.pwd.length.user	0-32	2	

License <license/>

This attribute's settings control aspects of the feature licensing system.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
license.polling.time	00:00 – 23:59	2:00am	The time to check whether or not the license has expired.

Provisioning <prov/>

This attribute's settings control aspects of the phone's boot server provisioning system.

Attribute	Permitted Values	Default	Interpretation
prov.fileSystem.rfs0.minFreeSpace	5-512	5	Minimum free space in Kbytes to reserve in the file system when downloading files from the boot server. Note: <i>Polycom recommends that you do not change these parameters.</i> Note: <i>For the SoundPoint IP 650 platform, <code>prov.fileSystem.ffi0.8meg.minFreeSpace</code> is internally replaced by 2X the value.</i>
prov.fileSystem.ffi0.4meg.minFreeSpace		420	
prov.fileSystem.ffi0.2meg.minFreeSpace		48	
prov.fileSystem.ffi0.8meg.minFreeSpace		512	
prov.polling.enabled	0, 1	0	If set to 1, automatic periodic boot server polling for upgrades is enabled.
prov.polling.mode	abs, rel	abs	Polling mode is <i>absolute</i> or <i>relative</i> .
prov.polling.period	integer greater than 3600	86400	Polling period in seconds. Rounded up to the nearest number of days in <i>abs</i> mode. Measured relative to boot time in <i>rel</i> mode.
prov.polling.time	Format is hh:mm	03:00	Only used in <i>abs</i> mode. Polling time.

RAM Disk <ramdisk/>

This attribute's settings control the phone's internal RAM disk feature.



Polycom recommends that you do not change these values.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
ramdisk.enable	0, 1	1	If set to 1, RAM disk will be available. The RAM disk is used to cache downloaded wave files, and other resources for the user interface.
ramdisk.bytesPerBlock	0, 32, 33, ..., 1024	0	These three parameters use internal defaults when value is set to 0. Note: For the SoundPoint IP 650 platform, <i>ramdisk.bytesPerBlock</i> is internally replaced by 2X the value.
ramdisk.blocksPerTrack	0, 1, 2, ..., 65536	0	
ramdisk.nBlocks	0, 1, 2, ..., 65536	4096	
ramdisk.minsize	50 to 16384	50	Smallest size in Kbytes of RAM disk to create before returning an error. RAM disk size is variable depending on the amount of device memory.
ramdisk.minfree	512 to 16384	3072	Minimum amount of free space that must be left after the RAM disk has been created. The RAM disk's size will be reduced as necessary in order to leave this amount of free RAM.

Request <request/>

This attribute includes:

- Delay <delay/>

Delay <delay/>

These settings control the phone's behavior when a request for restart or reconfiguration is received.

Attribute	Permitted Values	Default	Interpretation
request.delay.type	Null, "audio", or "call"	call	Defines the strategy to adopt before a request gets executed. If set to "audio", a request can be executed as soon as there is no active audio on the phone, independently of any call state. If set to "call", a request can be executed as soon as there are no calls in any state on the phone.

Feature <feature/>

These settings control the activation or deactivation of a feature at run time. In the table below, *x* is the feature number.

Attribute	Permitted Values	Interpretation
feature.x.name	"presence", "messaging", "directory", "calllist", "ring-download", "calllist-received", "calllist-placed", "calllist-missed", "url-dialing", "call-park", "group-call-pickup", "directed-call-pickup", "last-call-return", "acd-login-logout", "acd-agent-available", "nway-conference", "call-recording", "corporate-directory"	These are features offered on the phone: "presence" is the presence feature including management of buddies and own status "messaging" is the instant messaging feature "directory" is the local directory feature "calllist" is the locally controlled call lists "ring-download" is run-time downloading of ringers "calllist-received" is the received-calls list feature (the "calllist" feature must be enabled for this feature to be available) "calllist-placed" is the placed-calls list feature (the "calllist" feature must be enabled for this feature to be available) "calllist-missed" is the missed-calls list feature (the "calllist" feature must be enabled for this feature to be available) "url-dialing" controls whether URL/name dialing is available from a private line (it is never available from a shared line) "call-park" is the call park and park-retrieve features "group-call-pickup" is the group call pickup feature "directed-call-pickup" is the directed call pickup feature "last-call-return" is the last call return feature "acd-login-logout" is the ACD login/logout feature "acd-agent-available" is the ACD agent available/unavailable feature "nway-conference" is the conference managing feature "call-recording" is the call recording and playback feature "corporate-directory" is the corporate directory feature

Attribute	Permitted Values	Interpretation
feature.x.enabled	0 or 1 (default) except for x=9	<p>If set to 0, the feature will be disabled.</p> <p>If set to 1, the feature will be enabled and usable by the local user.</p> <p>Note: The "url-dialing" feature must be disabled by setting <code>feature.9.enabled</code> to 0 in order to prevent unknown callers from being identified on the display by an IP address.</p> <p>Note: The "call list" feature can be disabled on all SoundPoint IP and SoundStation IP platforms except the SoundPoint IP 330/320.</p> <p>Note: For <code>feature.16.name = "nway-conference"</code>:</p> <ul style="list-style-type: none"> • If set to 0, the n-way conferencing feature is disabled, meaning that three-way conferencing can exist, but there is no manage conference page. • If set to 1, the n-way conferencing feature is enabled, the maximum number of conference parties for the platform can exist, and there is a manage conference page. <p>Note: The manage conference feature is always disabled on the SoundPoint IP 301, 320/330, 430, 501 600/601 phone.</p> <p>Note: <code>feature.16.name = "nway-conference"</code>, <code>feature.17.name = "call-recording"</code>, and <code>feature.19.name = "corporate-directory"</code> are charged for separately. To activate these features, you must go to the Polycom Resource Center (http://extranet.polycom.com/csnprod/signon.html) to retrieve the activation code.</p>

Resource <res/>

This attribute's settings control the maximum size of an external resource retrieved at run time.

This attribute also includes:

- [Finder <finder/>](#)
- [Quotas <quotas/>](#)

Finder <finder/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
res.finder.sizeLimit	positive integer	300	If a resource that is being downloaded to the phone is larger than this value * 1024 bytes (= the maximum size), the resource will be automatically truncated to the maximum size defined. Note: For the SoundPoint IP 650 platform, this value is internally replaced by 2X the value.
res.finder.minfree	1 to 2048	600	A resource will not be downloaded to the phone if the amount of free memory is less than this value * 1024 bytes (= the minimum size). This parameter is used for 16MB SDRAM platforms and scaled up for platforms with more SDRAM. If set to 0 or Null, the default value of 600 is used. Note: For the SoundPoint IP 650 platform, this value is internally replaced by 2X the value.

Quotas <quotas/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Interpretation
res.quotas.x.name	1="tone", 2="bitmap", 3="font", 5="background"	The name of the sub-application for which the particular quota will apply: "tone" relates to all downloaded tones and sound effects "bitmap" relates to all downloaded bitmaps "font" relates to all downloaded fonts "background" relates to all downloaded backgrounds
res.quotas.x.value	positive integer	When a particular resource (one of category "font", "bitmap", or "font") is downloaded to the phone, a quota equal to this value * 1024 bytes of compound data size is applied for that category. If downloading a resource would exceed the quota for that category, the resource will not be downloaded and a predefined default will be used instead. For res.quotas.x.value, the default is 600 KB for tones, 10 KB for bitmaps and fonts, and 600KB for backgrounds. Note: For the SoundPoint IP 650 platform, this value is internally replaced by 2X the value.

Microbrowser <mb/>

This attribute's settings control the home page, proxy and size limits to be used by the Microbrowser when it is selected to provide services. The Microbrowser is supported on the SoundPoint IP 430, 501, 550, 560, 601, and 650 and the SoundStation IP 4000 phones.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
mb.proxy	Null or domain name or IP address in the format <address>:<port>	Null. Default port = 8080	Address of the desired HTTP proxy to be used by the Microbrowser. If blank, normal unproxied HTTP is used by the Microbrowser.

This attribute also includes:

- [Idle Display <idleDisplay/>](#)
- [Main Browser <main/>](#)
- [Browser Limits <limits/>](#)

Idle Display <idleDisplay/>

The Microbrowser can be used to create a display that will be part of the phone's idle display. These settings control the home page and the refresh rate.

Attribute	Permitted Values	Default	Interpretation
mb.idleDisplay.home	Null or any fully formed valid HTTP URL. Length up to 255 characters.	Null	URL used for Microbrowser idle display home page. For example: <code>http://www.example.com/xhtml/frontpage.cgi?page=home</code> . If empty, there will be no Microbrowser idle display feature. Note that the Microbrowser idle display will displace the idle display indicator (refer to <code>ind.idleDisplay.enabled</code> in Indicators <ind/> on page A-71). Note: If <code>ind.idleDisplay.enabled</code> is enabled, miscellaneous XML errors can occur on SoundPoint IP 430, 501, 550, 560, 600, 601, and 650 and SoundStation IP 4000 phones.

Attribute	Permitted Values	Default	Interpretation
mb.idleDisplay.refresh	0 or an integer > 5	0	<p>The period in seconds between refreshes of the idle display Microbrowser's content. If set to 0, the idle display Microbrowser is not refreshed. The minimum refresh period is 5 seconds (values from 1 to 4 are ignored, and 5 is used).</p> <p>Note: If an HTTP Refresh header is detected, it will be respected, even if this parameter is set to 0. The refresh parameter will be respected only in the event that a refresh fails. Once a refresh is successful, the value in the HTTP refresh header, if available, will be used.</p>

Main Browser <main/>

This setting controls the home page used by the Microbrowser when that function is selected.

Attribute	Permitted Values	Default	Interpretation
mb.main.home	Any fully formed valid HTTP URL. Length up to 255 characters.	Null	<p>URL used for Microbrowser home page. If blank, the browser will notify the user that a blank home-page was used.</p> <p>For example: http://www.example.com/xhtml/frontpage.cgi?page=home.</p>
mb.main.statusbar	0, 1	Null	<p>Flag to determine whether or not to turn off display of status messages.</p> <p>If set to 1, the display of the status bar is enabled.</p> <p>If set to 0, or Null, the display of the status bar is disabled.</p>
mb.main.idleTimeout	0 - 600, seconds	Null	<p>Timeout for the interactive browser. If the interactive browser remains idle for a defined period of time, the phone should return to the idle browser.</p> <p>If set to 0, there is no timeout.</p> <p>If set to Null, the value from <code>up.idleTimeout</code> is used. Refer to User Preferences <up/> on page A-23. If <code>mb.main.idleTimeout</code> and <code>up.idleTimeout</code> are Null, the timeout is 20 seconds.</p> <p>If set to value greater than 0 and less than 600, the timeout is for that number of seconds.</p>

Browser Limits <limits/>

These settings limit the size of object which the Microbrowser will display by limiting the amount of memory available for the Microbrowser.

Attribute	Permitted Values	Default	Interpretation
mb.limits.nodes	positive integer	256	Limits the number of tags that the XML parser will handle. This limits the amount of memory used by complicated pages. A maximum total of 500 (256 each) is recommended. This value is used as referent values for 16MB of SDRAM. Note: Increasing this value may have a detrimental effect on performance of the phone.
mb.limits.cache	positive integer	200	Limits the total size of objects downloaded for each page (both XHTML and images). Once this limit is reached, no more images are downloaded until the next page is requested. Units = kBytes. This value is used as referent values for 16MB of SDRAM. Note: Increasing this value may have a detrimental effect on performance of the phone.

Per-Phone Configuration

This section covers the parameters in the per-phone example configuration file **phone1.cfg**. This file would normally be used as a template for the per-phone configuration files. For more information, refer to [Deploying Phones From the Boot Server](#) on page 3-15.



Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first. For more information, refer to the "Configuration File Management on SoundPoint IP Phones" whitepaper at www.polycom.com/support/voice/.

The parameters include:

- [Registration <reg/>](#)
- [Calls <call/>](#)
- [Diversion <divert/>](#)
- [Dial Plan <dialplan/>](#)
- [Messaging <msg/>](#)
- [Network Address Translation <nat/>](#)
- [Attendant <attendant/>](#)
- [Roaming Buddies <roaming_buddies/>](#)
- [Roaming Privacy <roaming_privacy/>](#)
- [User Preferences <user_preferences/>](#)

Registration <reg/>

SoundPoint IP 301, 320, 330, and 430 support a maximum of two unique registrations, SoundPoint IP 501 supports three, the SoundPoint IP 550 and 560 supports four, and SoundPoint IP 600, 601, and 650 support six. Up to three SoundPoint IP Expansion Modules can be added to a single host SoundPoint IP 601 and 650 phone increasing the total number of buttons to 12 registrations on the IP 601 and 34 registrations on the IP 650. Each registration can optionally be associated with a private array of servers for completely segregated signaling. The SoundStation IP 4000 supports a single registration.

In the following table, x is the registration number. IP 301, 320, 330, 430: x=1-2;
 IP 501: x=1-3; IP 550, 560: x=1-4; IP 600: x=1-6; IP 601: x=1-12;
 IP 650: x=1-34; IP 4000: x=1.

Attribute	Permitted Values	Default	Interpretation
reg.x.csta	0, 1	Null	If set to 1, uaCSTA is enabled. If reg.x.csta is not Null, this attribute overrides the global CSTA flag in the sip.cfg configuration file.
reg.x.displayName	UTF-8 encoded string	Null	Display name used for local user interface as well as SIP signaling.
reg.x.address	string in the format userPart from userPart@domain	Null	The user part or the user and the host part of the phone's SIP URI. The user part of the phone's SIP URI. For example, reg.x.address="1002" from 1002@polycom.com or reg.x.address="1002@polycom.com".
reg.x.label	UTF-8 encoded string	Null	Text label to appear on the display adjacent to the associated line key. If omitted, the label will be derived from the user part of reg.x.address.
reg.x.lcs	0, 1	0	If set to 1, the Microsoft Live Communications Server is supported for registration x.
reg.x.type	private OR shared	private	If set to private, use standard call signaling. If set to shared, augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.
reg.x.thirdPartyName	string in the same format as reg.x.address	Null	This field must match the reg.x.address value of the other registration which makes up the bridged line appearance (BLA). It must be Null in all other cases.
reg.x.auth.userId	string	Null	User ID to be used for authentication challenges for this registration. If non-Null, will override the "Reg User x" parameter entered into the Authentication submenu off of the Settings menu on the phone.
reg.x.auth.password	string	Null	Password to be used for authentication challenges for this registration. If non-Null, will override the "Reg Password x" parameter entered into the Authentication submenu off of the Settings menu on the phone.

Attribute	Permitted Values	Default	Interpretation
reg.x.server.y.address	dotted-decimal IP address or host name	Null	<p>Optional IP address or host name, port, transport, registration period, fail-over parameters and linesize subscription period of a SIP server that accepts registrations. Multiple servers can be listed starting with y=1, 2, ... for fault tolerance. If specified, these servers may override the servers specified in sip.cfg in <code>Server <server/></code> on page A-7.</p> <p>Note: If the <code>reg.x.server.y.address</code> parameter is non-Null, <u>all</u> of the <code>reg.x.server.y.xxx</code> parameters will override the parameters specified in sip.cfg in <code>Server <server/></code> on page A-7.</p> <p>Note: If the <code>reg.x.server.y.address</code> parameter is non-Null, it takes precedence even if the DHCP server is available.</p> <p>Note: TLS is not supported on SoundPoint IP 300 and 500 phones.</p>
reg.x.server.y.port	0, Null, 1 to 65535	Null	
reg.x.server.y.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	
reg.x.server.y.expires	positive integer	Null	
reg.x.server.y.register	0, 1	Null	
reg.x.server.y.expires.overlap	positive integer, minimum 5, maximum 65535	60	
reg.x.server.y.retryTimeOut	Null or non-negative integer	Null	
reg.x.server.y.retryMaxCount	Null or non-negative integer	Null	
reg.x.server.y.expires.lineSeize	positive integer	Null	
reg.x.server.y.lcs	0, 1	0	This attribute overrides the <code>reg.x.lcs</code> . If set to 1, the Microsoft Live Communications Server is supported for registration x.
reg.x.acd-login-logout	0, 1	0	If both parameters are set to 1 for a registration, the ACD feature will be enabled for that registration.
reg.x.acd-agent-available	0, 1	0	
reg.x.ringType	1 to 22	2	The ringer to be used for calls received by this registration. Default is the first non-silent ringer.
reg.x.lineKeys	1 to max	1	<p><i>max</i> = the number of line keys on the phone. <i>max</i> = 1 on SoundStation IP 4000 <i>max</i> = 2 on IP 301, 320, 330, 430, <i>max</i> = 3 on IP 501, <i>max</i> = 4 on IP 550, 560, <i>max</i> = 6 on IP 600, <i>max</i> = 48 on IP 601, 650 (without any Expansion Modules attached, only 6 line keys are available)</p> <p>The number of line keys on the phone to be associated with registration 'x'.</p>

Attribute	Permitted Values	Default	Interpretation
reg.x.callsPerLineKey	1 to 24 OR 1 to 8	24 OR 8	<p>For the SoundPoint IP 600, 601, and 650 the permitted range is 1 to 24 and the default is 24.</p> <p>For all other phones the permitted range is 1 to 8 and the default is 8.</p> <p>This is the number of calls or conferences which may be active or on hold per line key associated with this registration.</p> <p>Note that this overrides <code>call.callsPerLineKey</code> for this registration. Refer to Call Handling Configuration <call/> on page A-58.</p>
reg.x.bargeInEnabled	0, 1	Null	<p>Allow remote user of SCA to interrupt call. (Works in a similar way to resume.)</p> <p>If set to 1, barge-in is enabled for line x.</p> <p>If set to 0 or Null, barge-in is disabled for line x.</p>
reg.x.outboundProxy.address	dotted-decimal IP address or host name	Null	<p>IP address or host name and port of a SIP server to which the phone shall send all requests.</p>
reg.x.outboundProxy.port	1 to 65535	5060	
reg.x.outboundProxy.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	<p>If set to Null or DNSnaptr: If reg.x.outboundProxy.address is a hostname and reg.x.outboundProxy.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.outboundProxy.address is an IP address, or a port is given, then UDP is used.</p> <p>If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails.</p> <p>If set to UDPOnly: Only UDP will be used.</p> <p>If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.</p> <p>If set to TCPOnly: Only TCP will be used.</p> <p>NOTE: TLS is not supported on SoundPoint IP 300 and 500 phones.</p>
reg.x.proxyRequire	string	Null	<p>The string that needs to appear in the "Proxy-Require" header. If Null, no "Proxy-Require" will be sent.</p>

Attribute	Permitted Values	Default	Interpretation
reg.x.serverFeatureControl.cf	0, 1	0	<p>If set to 1, server-based call forwarding is enabled. The call server has control of call forwarding.</p> <p>If set to 0, server-based call forwarding is not enabled. This is the old behavior.</p> <p>If <code>reg.x.serverFeatureControl.cf</code> is not Null, this attribute overrides the global server-based call forwarding flag in the sip.cfg configuration file.</p>
reg.x.serverFeatureControl.dnd	0, 1	0	<p>If set to 1, server-based DND is enabled. The call server has control of DND.</p> <p>If set to 0, server-based DND is not enabled. This is the old behavior.</p> <p>If <code>reg.x.serverFeatureControl.dnd</code> is not Null, this attribute overrides the global server-based call forwarding flag in the sip.cfg configuration file.</p>
reg.x.auth.optimizedInFailover	0, 1	0	<p>If set to 1, when failover occurs, the first new SIP request is sent to the server that sent the proxy authentication request.</p> <p>If set to 0, when failover occurs, the first new SIP request is sent to the server with the highest priority in the server list.</p> <p>If this parameter is Null, <code>voIpProt.SIP.authOptimizedInFailover</code> is checked.</p> <p>If both parameters are set, this parameter takes precedence.</p>

Calls <call/>

This attribute affects the call-oriented per-phone configuration.

This attribute includes:

- [Do Not Disturb <donotdisturb/>](#)
- [Automatic Off-Hook Call Placement <autoOffHook/>](#)
- [Missed Call Configuration <serverMissedCall/>](#)
- [Missed Call Tracking <missedCallTracking/>](#)
- [Call Waiting <callWaiting/>](#)

Do Not Disturb <donotdisturb/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.donotdisturb.perReg	0, 1	0	If set to 1, the DND feature will allow selection of DND on a per-registration basis.

Automatic Off-Hook Call Placement <autoOffHook/>

An optional per-registration feature is supported which allows automatic call placement when the phone goes off-hook.

In the following table, x is the registration number. IP 301, 320, 330, 430: x=1-2; IP 501: x=1-3; IP 550, 560: x=1-4; IP 600: x=1-6; IP 601: x=1-12; IP 650: x=1-34; IP 4000: x=1.

Attribute	Permitted Values	Default	Interpretation
call.autoOffHook.x.enabled	0, 1	0	If set to 1, a call will be automatically placed to the contact specified upon going off-hook on this registration.
call.autoOffHook.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	

Missed Call Configuration <serverMissedCall/>

The phone supports a per-registration configuration of which events will cause the locally displayed "missed calls" counter to be incremented.

In the following table, x is the registration number. IP 301, 320, 330, 430: x=1-2; IP 501: x=1-3; IP 550, 560: x=1-4; IP 600: x=1-6; IP 601: x=1-12; IP 650: x=1-34; IP 4000: x=1.

Attribute	Permitted Values	Default	Interpretation
call.serverMissedCall.x.enabled	0, 1	0	If set to 0, all missed-call events will increment the counter. If set to 1, only missed-call events sent by the server will increment the counter.

Missed Call Tracking <missedCallTracking/>

You can enable/disable missed call tracking on a per-line basis.

In the following table, x is the registration number. IP 301, 320, 330, 430: x=1-2;
IP 501: x=1-3; IP 550, 560: x=1-4; IP 600: x=1-6; IP 601: x=1-12;
IP 650: x=1-34; IP 4000: x=1.

Attribute	Permitted Values	Default	Interpretation
call.missedCallTracking.x.enabled	0, 1	1	<p>If set to 1 or Null, missed call tracking is enabled.</p> <p>If call.missedCallTracking.x.enabled is set to 0, then missedCall counter is not updated regardless of what call.serverMissedCalls.x.enabled is set to (and regardless of how the server is configured). There is no Missed Call List provided under Menu > Features of the phone.</p> <p>If call.missedCallTracking.x.enabled is set to 1 and call.serverMissedCalls.x.enabled is set to 0, then the number of missedCall counter is incremented regardless of how the server is configured.</p> <p>If call.missedCallTracking.x.enabled is set to 1 and call.serverMissedCalls.x.enabled is set to 1, then the handling of missedCalls depends on how the server is configured.</p>

Call Waiting <callWaiting/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.callWaiting.ring	beep, ring, silent	beep	<p>Specifies the ring tone heard on an incoming call when another call is active.</p> <p>If set to Null, the default value is beep.</p>

Diversion <divert/>

The phone has a flexible call forward/diversion feature for each registration. In all cases, a call will only be diverted if a non-Null contact has been configured.

In the following tables, x is the registration number. IP 301, 320, 330, 430: x=1-2; IP 501: x=1-3; IP 550, 560: x=1-4; IP 600: x=1-6; IP 601: x=1-12; IP 650: x=1-34; IP 4000: x=1.

Attribute	Permitted Values	Default	Interpretation
divert.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	The forward-to contact used for all automatic call diversion features unless overridden by a specific contact of a per-call diversion feature (refer to below).
divert.x.autoOnSpecificCaller	0, 1	1	If set to 1, calls may be diverted using the Auto Divert feature of the directory. This is a global flag. Note: If server-based call forwarding is enabled, this parameter is disabled.
divert.x.sharedDisabled	0, 1	1	If set to 1, all diversion features on that line will be disabled if the line is configured as shared.

This attribute also includes:

- [Forward All <fwd/>](#)
- [Busy <busy/>](#)
- [No Answer <noanswer/>](#)
- [Do Not Disturb <dnd/>](#)

Forward All <fwd/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
divert.fwd.x.enabled	0, 1	1	If set to 1, the user will be able to enable universal call forwarding through the soft key menu. Note: If server-based call forwarding is enabled, this parameter is enabled.

Busy <busy/>

Calls can be automatically diverted when the phone is busy.

Attribute	Permitted Values	Default	Interpretation
divert.busy.x.enabled	0, 1	1	If set to 1, calls will be forwarded on busy to the contact specified below. Note: If server-based call forwarding is enabled, this parameter is disabled.
divert.busy.x.timeout	positive integer	60	Time in seconds to allow altering before initiating the diversion.
divert.busy.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Forward-to contact for calls forwarded due to busy status, if Null, divert.x.contact will be used.

No Answer <noanswer/>

The phone can automatically divert calls after a period of ringing.

Attribute	Permitted Values	Default	Interpretation
divert.noanswer.x.enabled	0, 1	1	If set to 1, calls will be forwarded on no answer to the contact specified. Note: If server-based call forwarding is enabled, this parameter is disabled.
divert.noanswer.x.timeout	positive integer	60	Time in seconds to allow altering before initiating the diversion.
divert.noanswer.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Forward-to contact used for calls forwarded due to no answer, if Null, divert.x.contact will be used.

Do Not Disturb <dnd/>

The phone can automatically divert calls when Do Not Disturb (DND) is enabled.

Attribute	Permitted Values	Default	Interpretation
divert.dnd.x.enabled	0, 1	0	If set to 1, calls will be forwarded on DND to the contact specified below. Note: If server-based DND or server-base call forwarding is enabled, this parameter is disabled.
divert.dnd.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Forward-to contact used for calls forwarded due to DND status, if Null <code>divert.x.contact</code> will be used.

Dial Plan <dialplan/>

Per-registration dial plan configuration is supported.

In the following tables, *x* is the registration number. IP 301, 320, 330, 430: *x*=1-2; IP 501: *x*=1-3; IP 550, 560: *x*=1-4; IP 600: *x*=1-6; IP 601: *x*=1-12; IP 650: *x*=1-34; IP 4000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.applyToCallListDial	0, 1	0	When present, and if <code>dialplan.x.digitmap</code> is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.
dialplan.x.applyToDirectoryDial	0, 1	0	When present, and if <code>dialplan.x.digitmap</code> is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.applyToUserDial	0, 1	1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.
dialplan.x.applyToUserSend	0, 1	1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.
dialplan.x.impossibleMatchHandling	0, 1 or 2	0	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.
dialplan.x.removeEndOfDial	0, 1	1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-16.

This attribute also includes:

- [Digit Map <digitmap/>](#)
- [Routing <routing/>](#)

[Digit Map <digitmap/>](#)

For more information on digit map syntax, refer to [Digit Map <digitmap/>](#) on page A-17.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.digitmap	A string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435; string is limited to 768 bytes and 30 segments; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on; '+' is allowed as a valid digit; extension letter 'R' is used as defined above.	Null	When present, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For more information, refer to Digit Map <digitmap/> on page A-17.
dialplan.x.digitmap.timeOut	string of positive integers separated by ' '	Null	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file. For more information, refer to Digit Map <digitmap/> on page A-17.

Routing <routing/>

This attribute allows specific routing paths for outgoing SIP calls to be configured independent of other 'default' configuration.

This attribute includes:

- [Server <server/>](#)
- [Emergency <emergency/>](#)

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.routing.server.y.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that will be used for routing calls. Multiple servers can be listed starting with y=1, 2, ... for fault tolerance.
dialplan.x.routing.server.y.port	1 to 65535	5060	

Emergency <emergency/>

In the following attributes, *y* is the index of the emergency entry description and *z* is the index of the server associated with the emergency entry *y*. For each emergency entry (index *y*), one or more server entry (indexes (*y,z*)) can be configured. *y* and *z* must both follow single step increasing numbering starting at 1.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.routing.emergency.y.value	Comma separated list of entries or single entry representing a or a combination of SIP URL.	Null Example: "15,17,18", "911", "sos".	This represents the URLs that should be watched for emergency routing. When one of these defined URL is detected as being dialed by the user, the call will be automatically directed to the defined emergency server.
dialplan.x.routing.emergency.y.server.z	positive integer	Null	Index representing the server defined in Server <server/> on page A-100 that will be used for emergency routing.

Messaging <msg/>

Message-waiting indication is supported on a per-registration basis.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
msg.bypassInstantMessage	0, 1	0	If set to 1, the display offering a choice of "Message Center" and "Instant Messages" will be bypassed when pressing the Messages key. The phone will act as if "Message Center" was chosen. Refer to Voice Mail Integration on page 4-36. Instant Messages will still be accessible from the Main Menu.

This attribute also includes:

- [Message Waiting Indicator <mw/>](#)

Message Waiting Indicator <mwI/>

In the following table, *x* is the registration number. IP 301, 320, 330, 430: *x*=1-2; IP 501: *x*=1-3; IP 550, 560: *x*=1-4; IP 600: *x*=1-6; IP 601: *x*=1-12; IP 650: *x*=1-34; IP 4000: *x*=1.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
msg.mwi.x.subscribe	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	If non-Null, the phone will send a SUBSCRIBE request to this contact after boot-up.
msg.mwi.x.callBackMode	contact or registration or disabled	"registration"	Configures message retrieval and notification for the line. If set to "contact", a call will be placed to the contact specified in the callback attribute when the user invokes message retrieval. If set to "registration", a call will be placed using this registration to the contact registered (the phone will call itself). If set to "disabled", message retrieval and message notification are disabled.
msg.mwi.x.callBack	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Contact to call when retrieving messages for this registration.

Network Address Translation <nat/>

These parameters define port and IP address changes used in NAT traversal. The port changes will change the port used by the phone, while the IP entry simply changes the IP advertised in the SIP signaling. This allows the use of simple NAT devices that can redirect traffic, but do not allow for port mapping. For example, port 5432 on the NAT device can be sent to port 5432 on an internal device, but not port 1234.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
nat.ip	dotted-decimal IP address	Null	IP address to advertise within SIP signaling - should match the external IP address used by the NAT device.
nat.signalPort	1024 to 65535	Null	If non-Null, this port will be used by the phone for SIP signaling, overriding the value set for <code>voIpProt.local.signalPort</code> in sip.cfg .
nat.mediaPortStart	1024 to 65535	Null	If non-Null, this attribute will be used to set the initially allocated RTP port, overriding the value set for <code>tcpIpApp.port.rtp.mediaPortRangeStart</code> in sip.cfg . Refer to RTP <rtsp/> on page A-56 .
nat.keepalive.interval	0 to 3600	Null	If non-Null (or 0), the keepalive interval in seconds. This parameter is used to set the interval at which phones will send a keep-alive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function as setup initially. The Microsoft Live Communications Server 2005 keepalive feature will override this interval. If you want to deploy phones behind a NAT and connect them to Live Communications Server, the keepalive interval received from the Live Communications Server must be short enough to keep the NAT port open. Once the TCP connection is closed, the phones stop sending keep-alive packets.

Attendant <attendant/>

Note

These attributes are available on SoundPoint IP 600, 601, and 650 phones (with an attached Expansion Module) only.

The Busy Lamp Field (BLF) / attendant console feature enhances support for a phone-based attendant console.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
attendant.uri	string	Null	For attendant console / busy lamp field (BLF) feature. This specifies the list SIP URI on the server. If this is just a user part, the URI is constructed with the server host name/IP.
attendant.reg	positive integer	1	For attendant console / BLF feature. This is the index of the registration which will be used to send a SUBSCRIBE to the list SIP URI specified in attendant.uri. For example, attendant.reg = 2 means the second registration will be used.

Roaming Buddies <roaming_buddies/>

Note

This attribute is used in conjunction with Microsoft Live Communications Server 2005 only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
roaming_buddies.reg	positive integer	Null	Specifies the line/registration number which has roaming buddies support enabled. If Null, roaming buddies is disabled. If value < 1, then value is replaced with 1. Warning: This parameter must be enabled (value > 0) if the call server is Microsoft Live Communications Server 2005.

Roaming Privacy <roaming_privacy/>

Note

This attribute is used in conjunction with Microsoft Live Communications Server 2005 only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
roaming_privacy.reg	positive integer	Null	Specifies the line/registration number which has roaming privacy support enabled. If Null, roaming privacy is disabled. If value < 1, then value is replaced with 1.

User Preferences <user_preferences/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
up.analogHeadsetOption	0, 1	0	Selects optional external hardware for use with a headset attached to the phone's analog headset jack. If set to 0, no compatible headset is attached. If set to 1, a DHSG-compatible headset is attached and can be used as an electronic hookswitch.

Flash Parameter Configuration

Any field in the bootROM setup menu and the application SIP Configuration menu can be set through a configuration file.

A DHCP server can be configured to point the phones to a boot server that has the required configuration files. The new settings will be downloaded by the phones and used to configure them. This removes the need for manual interaction with phones to configure basic settings. This is especially useful for initial installation of multiple phones.

These device settings are detected when the application starts. If the new settings would normally cause a reboot if they were changed in the application Network Configuration menu, then they will cause a reboot when the application starts.

Caution

The parameters for this feature should be put in separate configuration files to simplify maintenance. Do not add them to existing configuration files (such as **sip.cfg**). One new configuration file will be required for parameters that should apply to all phones, and individual configuration files will be required for phone-specific parameters such as SIP registration information.

The global `device.set` parameter must be enabled when the initial installation is done, and then it should be disabled. This prevents subsequent reboots by individual phones triggering a reset of parameters on the phone that may have been tweaked since the initial installation.

Caution

This feature is very powerful and should be used with caution. For example, an incorrect setting could set the IP Address of multiple phones to the same value. Note that some parameters may be ignored, for example if DHCP is enabled it will still override the value set with `device.net.ipAddress`.

Individual parameters are checked to see whether they are in range, however, the interaction between parameters is not checked. If a parameter is out of range, an error message will appear in the log file and parameter will not be used.

Incorrect configuration could cause phones to get into a reboot loop. For example, server A has a configuration file that specifies that server B should be used, which has a configuration file that specifies that server A should be used.

Polycom recommends that you test the new configuration files on two phones before initializing all phones. This should detect any errors including IP address conflicts.

This flash attributes are defined as follows:

Name	Possible Values	Description
device.set	0 or 1 default = 0	If set to 0, do not use any <code>device.xxx.yyy</code> fields to set any parameters. Set this to 0 after the initial installation. If set to 1, use the <code>device.xxx.yyy</code> fields that have <code>device.xxx.yyy.set = 1</code> . Set this to 1 for the initial installation only.
device.xxx.yyy.set	0 or 1 default = 0	If set to 0, do not use the <code>device.xxx.yyy</code> value. If set to 1, use the <code>device.xxx.yyy</code> value. For example, if <code>device.net.ipAddress.set = 1</code> , then use the contents of the <code>device.net.ipAddress</code> field.
device.net.ipAddress	dotted-decimal IP address	Phone's IP address. Note: This field is not used when DHCP client is enabled.
device.net.subnetMask	dotted-decimal IP address	Phone's subnet mask. Note: This field is not used when DHCP client is enabled.
device.net.IPgateway	dotted-decimal IP address	Phone's default router / IP gateway. Note: This field is not used when DHCP client is enabled.
device.net.vlanId	Null, 0 to 4094	Phone's 802.1Q VLAN identifier. Note: Null = no VLAN tagging
device.net.cdpEnabled	0 or 1	If set to 1, the phone will attempt to determine its VLAN ID through the CDP.
device.dhcp.enabled	0 or 1	For description, refer to DHCP or Manual TCP/IP Setup on page 3-2.

Name	Possible Values	Description
device.dhcp.offerTimeout	1 to 600	Number of seconds the phone waits for secondary DHCP Offer messages before selecting an offer.
device.dhcp.bootSrvUseOpt	0 to 3	For descriptions, refer to DHCP Menu on page 3-7.
device.dhcp.bootSrvOpt	128 to 254 (Cannot be the same as VLAN ID Option)	
device.dhcp.bootSrvOptType	0 or 1	
device.dhcp.dhcpVlanDiscUseOpt	0 to 2	
device.dhcp.dhcpVlanDiscOpt	128 to 254 (Cannot be the same as Boot Server Option)	
device.prov.serverName	any string	For descriptions, refer to Server Menu on page 3-9.
device.prov.serverType	0 to 4	
device.prov.user	any string	
device.prov.password	any string	
device.prov.appProvType	0 or 1	
device.prov.appProvString	any string	
device.prov.redunAttemptLimit	10, Null	
device.prov.redunInterAttemptDelay	300, Null	
device.snmp.serverName	any string	
device.snmp.gmtOffset	-43200 to 46800	GMT offset in seconds, corresponding to -12 to +13 hours.
device.dns.serverAddress	dotted-decimal IP address	Primary server to which the phone directs Domain Name System queries.
device.dns.altSrvAddress	dotted-decimal IP address	Secondary server to which the phone directs Domain Name System queries.
device.dns.domain	any string	The phone's DNS domain.
device.auth.localAdminPassword	any string	The phone's local administrator password.
device.auth.localUserPassword	any string	The phone user's local password.

Name	Possible Values	Description
device.auth.regUserx	any string	The SIP registration user name for registration x where x = 1 to 12.
device.auth.regPasswordx	any string	The SIP registration password for registration x where x = 1 to 12.
device.sec.configEncryption.key	any string	Configuration encryption key that is used for encryption of configuration files.
device.syslog.serverName	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.
device.syslog.transport	None=0, UDP=1, TCP=2, TLS=3	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
device.syslog.facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3165. The default value is 16, which maps to "local 0".
device.syslog.renderLevel	0 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on <code>log.render.level</code> and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page A-77.
device.syslog.prependMac	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.
device.em.power	Enabled, Disabled, Null	Refer to the EM Power parameter in Main Menu on page 3-6.
device.net.etherVlanFilter	Enabled, Disabled	Refer to the VLAN Filtering parameter in Ethernet Menu on page 3-11.
device.net.etherStormFilter	Enabled, Disabled	Refer to the Storm Filtering parameter in Ethernet Menu on page 3-11.
device.sec.SSL.certList	all, custom, default	The type of certificate list.
device.sec.SSL.customCert	X.509 certificate	The certificate value.

Session Initiation Protocol (SIP)

This chapter provides a description of the basic Session Initiation Protocol (SIP) and the protocol extensions that are supported by the current SIP application. To find the applicable Request For Comments (RFC) document, go to <http://www.ietf.org/rfc.html> and enter the RFC number.

This chapter contains information on:

- Basic Protocols – All the basic calling functionality described in the SIP specification is supported. Transfer is included in the basic SIP support.
- Protocol Extensions – Extensions add features to SIP that are applicable to a range of applications, including reliable 1xx responses and session timers.

For information on supported RFC's and Internet drafts, refer to the following section, [RFC and Internet Draft Support](#).

This chapter also describes:

- [Request Support](#)
- [Header Support](#)
- [Response Support](#)
- [Hold Implementation](#)
- [Reliability of Provisional Responses](#)
- [Transfer](#)
- [Third Party Call Control](#)
- [SIP for Instant Messaging and Presence Leveraging Extensions](#)
- [Shared Call Appearance Signaling](#)
- [Bridged Line Appearance Signaling](#)

RFC and Internet Draft Support

The following RFC's and Internet drafts are supported:

- RFC 1321 – The MD5 Message-Digest Algorithm
- RFC 2327 – SDP: Session Description Protocol
- RFC 2387 – The MIME Multipart / Related Content-type
- RFC 2976 – The SIP INFO Method
- RFC 3261 – SIP: Session Initiation Protocol (replacement for RFC 2543)
- RFC 3262 – Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263 – Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264 – An Offer / Answer Model with the Session Description Protocol (SDP)
- RFC 3265 – Session Initiation Protocol (SIP) - Specific Event Notification
- RFC 3311 – The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3325 – SIP Asserted Identity
- RFC 3515 – The Session Initiation Protocol (SIP) Refer Method
- RFC 3555 – MIME Type of RTP Payload Formats
- RFC 3611 – RTP Control Protocol Extended reports (RTCP XR)
- RFC 3665 – Session Initiation Protocol (SIP) Basic Call Flow Examples
- draft-ietf-sip-cc-transfer-05.txt – SIP Call Control - Transfer
- RFC 3725 – Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3842 – A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 3856 – A Presence Event Package for Session Initiation Protocol (SIP)
- RFC 3891 – The Session Initiation Protocol (SIP) “Replaces” Header
- RFC 3892 – The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3959 – The Early Session Disposition Type for the Session Initiation Protocol (SIP)
- RFC 3960 – Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- RFC 3968 – The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session Initiation Protocol (SIP)

- RFC 3969 – The Internet Assigned Number Authority (IANA) Uniform Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP)
- RFC 4028 – Session Timers in the Session Initiation Protocol (SIP)
- RFC 4235 – An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4662 – Session Initiation Protocol (SIP) Event Notification Extension for Resource Lists
- draft-levy-sip-diversion-04.txt – Diversion Indication in SIP
- draft-anil-sipping-bla-02.txt – Implementing Bridged Line Appearances (BLA) Using Session Initiation Protocol (SIP)
- draft-ietf-sip-privacy-04.txt – SIP Extensions for Network-Asserted Caller Identity and Privacy within Trusted Networks
- draft-levy-sip-diversion-06.txt – Diversion Indication in SIP
- draft-ietf-sipping-cc-conferencing-03.txt – SIP Call Control - Conferencing for User Agents
- draft-ietf-sipping-rtcp-summary-02.txt – Session Initiation Protocol Package for Voice Quality Reporting Event
- draft-ietf-sip-connect-reuse-04.txt – Connection Reuse in the Session Initiation Protocol (SIP)

Request Support

The following SIP request messages are supported:

Method	Supported	Notes
REGISTER	Yes	
INVITE	Yes	
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	
NOTIFY	Yes	
REFER	Yes	
PRACK	Yes	

Method	Supported	Notes
INFO	Yes	RFC 2976, the phone does not generate INFO requests, but will issue a final response upon receipt. No INFO message bodies are parsed.
MESSAGE	Yes	Final response is sent upon receipt. Message bodies of type text/plain are sent and received.
UPDATE	Yes	

Header Support

The following SIP request headers are supported:

Note

In the following table, a “Yes” in the Supported column means the header is sent and properly parsed.

Header	Supported	Notes
Accept	Yes	
Accept-Encoding	No	
Accept-Language	No	
Alert-Info	Yes	
Allow	Yes	
Allow-Events	Yes	
Authentication-Info	No	
Authorization	Yes	
Call-ID	Yes	
Call-Info	Yes	
Contact	Yes	
Content-Disposition	No	
Content-Encoding	No	
Content-Language	No	
Content-Length	Yes	
Content-Type	Yes	
CSeq	Yes	
Date	No	

Header	Supported	Notes
Diversion	Yes	
Error-Info	No	
Event	Yes	
Expires	Yes	
From	Yes	
In-Reply-To	No	
Max-Forwards	Yes	
Min-Expires	No	
Min-SE	Yes	
MIME-Version	No	
Organization	No	
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
Priority	No	
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	
Proxy-Require	No	
RAck	Yes	
Record-Route	Yes	
Refer-To	Yes	
Referred-By	Yes	
Remote-Party-ID	Yes	
Replaces	Yes	
Reply-To	No	
Require	Yes	
Retry-After	No	
Route	Yes	
RSeq	Yes	
Server	No	
Session-Expires	Yes	

Header	Supported	Notes
Subject	No	
Subscription-State	Yes	
Supported	Yes	
Timestamp	No	
To	Yes	
Unsupported	No	
User-Agent	Yes	
Via	Yes	
Warning	No	
WWW-Authenticate	Yes	

Response Support

The following SIP responses are supported:

Note

In the following table, a "Yes" in the Supported column means the header is sent and properly parsed. The phone may not actually generate the response.

1xx Responses - Provisional

Response	Supported	Notes
100 Trying	Yes	
180 Ringing	Yes	
181 Call Is Being Forwarded	No	
182 Queued	No	
183 Session Progress	Yes	

2xx Responses - Success

Response	Supported	Notes
200 OK	Yes	
202 Accepted	Yes	In REFER transfer.

3xx Responses - Redirection

Response	Supported	Notes
300 Multiple Choices	Yes	
301 Moved Permanently	Yes	
302 Moved Temporarily	Yes	
305 Use Proxy	No	
380 Alternative Service	No	

4xx Responses - Request Failure**Note**

All 4xx responses for which the phone does not provide specific support will be treated the same as 400 Bad Request.

Response	Supported	Notes
400 Bad Request	Yes	
401 Unauthorized	Yes	
402 Payment Required	No	
403 Forbidden	No	
404 Not Found	Yes	
405 Method Not Allowed	Yes	
406 Not Acceptable	No	
407 Proxy Authentication Required	Yes	
408 Request Timeout	No	
410 Gone	No	
413 Request Entity Too Large	No	
414 Request-URI Too Long	No	
415 Unsupported Media Type	Yes	
416 Unsupported URI Scheme	No	
420 Bad Extension	No	
421 Extension Required	No	
423 Interval Too Brief	No	
480 Temporarily Unavailable	Yes	

Response	Supported	Notes
481 Call/Transaction Does Not Exist	Yes	
482 Loop Detected	Yes	
483 Too Many Hops	No	
484 Address Incomplete	Yes	
485 Ambiguous	No	
486 Busy Here	Yes	
487 Request Terminated	Yes	
488 Not Acceptable Here	Yes	
491 Request Pending	No	
493 Undecipherable	No	

5xx Responses - Server Failure

Response	Supported	Notes
500 Server Internal Error	Yes	
501 Not Implemented	Yes	
502 Bad Gateway	No	
503 Service Unavailable	No	
504 Server Time-out	No	
505 Version Not Supported	No	
513 Message Too Large	No	

6xx Responses - Global Failure

Response	Supported	Notes
600 Busy Everywhere	No	
603 Decline	Yes	
604 Does Not Exist Anywhere	No	
606 Not Acceptable	No	

Hold Implementation

The phone supports both currently accepted means of signaling hold.

The first method, no longer recommended due in part to the RTCP problems associated with it, is to set the “c” destination addresses for the media streams in the SDP to zero, for example, c=0.0.0.0.

The second, and preferred, method is to signal the media directions with the “a” SDP media attributes sendonly, recvonly, inactive, or sendrecv. The hold signaling method used by the phone is configurable (refer to [SIP <SIP/>](#) on page [A-10](#)), but both methods are supported when signaled by the remote end point.

Note

Even if the phone is set to use c=0.0.0.0, it will not do so if it gets any sendrecv, sendonly, or inactive from the server. These flags will cause it to revert to the other hold method.

Reliability of Provisional Responses

The phone fully supports RFC 3262 - *Reliability of Provisional Responses*.

Transfer

The phone supports transfer using the REFER method specified in draft-ietf-sip-cc-transfer-05 and RFC 3515.

Third Party Call Control

The phone supports the delayed media negotiations (INVITE without SDP) associated with third party call control applications.

When used with an appropriate server, the User Agent Computer Supported Telecommunications Applications (uaCSTA) feature on the phone may be utilized for remote control of the phone from computer applications such as Microsoft Office Communicator.

The phone is compliant with “Using CSTA for SIP Phone User Agents (uaCSTA), ECMA TR/087” for the Answer Call, Hold Call, and Retrieve Call functions and “Services for Computer Supported Telecommunications Applications Phase III”, ECMA - 269 for the Conference Call function.

This feature is enabled by configuration parameters described in [SIP <SIP/>](#) on page [A-10](#) and [Registration <reg/>](#) on page [A-89](#) and needs to be activated by a feature application key.

SIP for Instant Messaging and Presence Leveraging Extensions

- The phone is compatible with the Presence and Instant Messaging features of Microsoft Windows Messenger 5.1.

Shared Call Appearance Signaling

A shared line is an address of record managed by a call server. The server allows multiple end points to register locations against the address of record.

The phone supports shared call appearances (SCA) using the SUBSCRIBE-NOTIFY method in the "SIP Specific Event Notification" framework (RFC 3265). The events used are:

- "call-info" for call appearance state notification
- "line-seize" for the phone to ask to seize the line

Bridged Line Appearance Signaling

A bridged line is an address of record managed by a server. The server allows multiple end points to register locations against the address of record.

The phone supports bridged line appearances (BLA) using the SUBSCRIBE-NOTIFY method in the "SIP Specific Event Notification" framework (RFC 3265). The events used are:

- "dialog" for bridged line appearance subscribe and notify

Miscellaneous Administrative Tasks

This appendix provides information required by varied aspects of the Session Initiation Protocol (SIP) application. This includes:

- [Trusted Certificate Authority List](#)
- [Encrypting Configuration Files](#)
- [Adding a Background Logo](#)
- [BootROM/SIP Application Dependencies](#)
- [Multiple Key Combinations](#)
- [Default Feature Key Layouts](#)
- [Assigning a VLAN ID Using DHCP](#)
- [Parsing Vendor ID Information](#)

Trusted Certificate Authority List

The following certificate authorities are trusted by the phone by default:

- ABAecom (sub., Am. Bankers Assn.) Root CA
- ANX Network CA by DST
- American Express CA
- American Express Global CA
- BelSign Object Publishing CA
- BelSign Secure Server CA
- Deutsche Telekom AG Root CA
- Digital Signature Trust Co. Global CA 1
- Digital Signature Trust Co. Global CA 2

- Digital Signature Trust Co. Global CA 3
- Digital Signature Trust Co. Global CA 4
- Entrust Worldwide by DST
- Entrust.net Premium 2048 Secure Server CA
- Entrust.net Secure Personal CA
- Entrust.net Secure Server CA
- Equifax Premium CA
- Equifax Secure CA
- GTE CyberTrust Global Root
- GTE CyberTrust Japan Root CA
- GTE CyberTrust Japan Secure Server CA
- GTE CyberTrust Root 2
- GTE CyberTrust Root 3
- GTE CyberTrust Root 4
- GTE CyberTrust Root 5
- GTE CyberTrust Root CA
- GlobalSign Partners CA
- GlobalSign Primary Class 1 CA
- GlobalSign Primary Class 2 CA
- GlobalSign Primary Class 3 CA
- GlobalSign Root CA
- National Retail Federation by DST
- TC TrustCenter, Germany, Class 1 CA
- TC TrustCenter, Germany, Class 2 CA
- TC TrustCenter, Germany, Class 3 CA
- TC TrustCenter, Germany, Class 4 CA
- Thawte Personal Basic CA
- Thawte Personal Freemail CA
- Thawte Personal Premium CA
- Thawte Premium Server CA

- Thawte Server CA
- Thawte Universal CA Root
- UPS Document Exchange by DST
- ValiCert Class 1 VA
- ValiCert Class 2 VA
- ValiCert Class 3 VA
- VeriSign Class 4 Primary CA
- Verisign Class 1 Public Primary Certification Authority
- Verisign Class 1 Public Primary Certification Authority - G2
- Verisign Class 1 Public Primary Certification Authority - G3
- Verisign Class 2 Public Primary Certification Authority
- Verisign Class 2 Public Primary Certification Authority - G2
- Verisign Class 2 Public Primary Certification Authority - G3
- Verisign Class 3 Public Primary Certification Authority
- Verisign Class 3 Public Primary Certification Authority - G2
- Verisign Class 3 Public Primary Certification Authority - G3
- Verisign Class 4 Public Primary Certification Authority - G2
- Verisign Class 4 Public Primary Certification Authority - G3
- Verisign/RSA Commercial CA
- Verisign/RSA Secure Server CA

Encrypting Configuration Files

The phone can recognize encrypted files, which it downloads from the boot server and it can encrypt files before uploading them to the boot server. There must be an encryption key on the phone to perform these operations. Configuration files (excluding the master configuration file), contact directories, and configuration override files can be encrypted.

A separate SDK, with a readme file, is provided to facilitate key generation and configuration file encryption and decrypt on a UNIX or Linux server. The utility is distributed as source code that runs under the UNIX operating system. For more information, contact Polycom Technical Support.

A key is generated by the utility and must be downloaded to the phone so that it can decrypt the files that were encrypted on the server. The `device.sec.configEncryption.key` configuration file parameter is used to set the key on the phone. The utility generates a random key and the encryption is Advanced Encryption Standard (AES) 128 in Cipher Block Chaining (CBC) mode. An example key would look like this:

```
Crypt=1;KeyDesc=companyNameKey1;Key=06a9214036b8a15b512e03d534120006;
```

If the phone doesn't have a key, it must be downloaded to the phone in plain text (a potential security hole if not using HTTPS). If the phone already has a key, a new key can be downloaded to the phone encrypted using the old key (refer to [Changing the Key on the Phone](#) on page C-5). At a later date, new phones from the factory will have a key pre-loaded in them. This key will be changed at regular intervals to enhance security.

It is recommended that all keys have unique descriptive strings in order to allow simple identification of which key was used to encrypt a file. This makes boot server management easier.

After encrypting a configuration file, it is useful to rename the file to avoid confusing it with the original version, for example rename `sip.cfg` to `sip.enc`. However, the directory and override filenames cannot be changed in this manner.

You can check whether an encrypted file is the same as an unencrypted file by:

1. Run the `configFileEncrypt` utility on the unencrypted file with the `"-d"` option. This shows the "digest" field.
2. Look at the encrypted file using WordPad and check the first line that shows a "Digest=..." field. If the two fields are the same, then the encrypted and unencrypted file are the same.

Note

If a phone downloads an encrypted file that it cannot decrypt, the action is logged, an error message displays, and the phone reboots. The phone will continue to do this until the boot server provides an encrypted file that can be read, an unencrypted file, or the file is removed from the master configuration file list.

Note

The SoundPoint IP 300 and 500 phones will always fail at decrypting files. These phones will recognize that a file is encrypted, but cannot decrypt it and will display an error. This information is logged. Encrypted configuration files can only be decrypted on the SoundPoint IP 301, 320, 330, 430, 501, 550, 560, 600, 601, and 650 and the SoundStation IP 4000 phones.

The master configuration file cannot be encrypted on the boot server. This file is downloaded by the bootROM that does not recognize encrypted files. For more information, refer to [Master Configuration Files](#) on page 2-5.

The following configuration file changes are required to modify this feature:

Central (boot server)	Configuration File: sip.cfg	Specify the phone-specific contact directory and the phone-specific configuration override file. For more information, refer to Encryption <encryption/> on page A-80 .
	Configuration file: <device>.cfg	Change the encryption key. For more information, refer to Flash Parameter Configuration on page A-105 .

Changing the Key on the Phone

For security purposes, it may be desirable to change the key on the phones and the server from time to time.

To change a key:

1. Put the new key into a configuration file that is in the list of files downloaded by the phone (specified in **000000000000.cfg** or **<Ethernet address>.cfg**).

Use the `device.sec.configEncryption.key` parameter to specify the new key.

2. Manually reboot the phone so that it will download the new key. The phone will automatically reboot a second time to use the new key.

At this point, the phone expects all encrypted configuration files on the boot server to use the new key and it will continue to reboot until this is the case. The files on the server must be updated to the new key or they must be made available in unencrypted format. Updating to the new key requires decrypting the file with the old key, then encrypting it with the new key.

Note that configuration files, contact directory files and configuration override files may all need to be updated if they were already encrypted. In the case of configuration override files, they can be deleted from the boot server so that the phone will replace them when it successfully boots.

Adding a Background Logo

This section provides instructions on how to add a background logo to all SoundPoint IP phones in your organization. You must be running at least BootROM 2.x.x and SIP 1.x.x. One bitmap file is required for each model, but SoundPoint IP 301 phones do not support bitmap logos.

Model	Width	Height	Color Depth
IP 301	n/a	n/a	n/a
IP 320/330	102	23	monochrome
IP 430	94	23	monochrome
IP 501	114	51	2-bit grayscale or monochrome
IP 600/601	209	109	2-bit grayscale or monochrome
IP 550/560/650	209	109	4-bit grayscale or monochrome
IP 4000	150	33	monochrome

Logos smaller than described in the table above are acceptable, but larger logos may be truncated or interfere with other areas of the user interface.

The SoundPoint IP 501/600/601 phones only support the four colors black, dark gray, light gray, and white. Any other colors will be approximated.

Color	RGB Values (Decimal)	RGB Values (Hexadecimal)
Black	0,0,0	00,00,00
Dark Gray	96,96,96	60,60,60
Light Gray	160,160,160	A0,A0,A0
White	255,255,255	FF,FF,FF

The SoundPoint IP 550/560/650 phones support a 4-bit grayscale, which is a smooth gradient from black (0, 0, 0) to white (FF, FF, FF).

The SoundStation IP 4000 phone only supports black and white. Any other colors will be rendered as either black or white.

Configuration File Changes

In the `<bitmaps>` section of `sip.cfg`, find the end of each model's bitmap list and add your bitmap to the end; do not include the `.bmp` extension:

```
<bitmaps>
  <IP_300 ... />
  <IP_330 ... bitmap.IP_330.66.name="logo-330" />
  <IP_500 ... bitmap.IP_500.66.name="logo-500" />
  <IP_600 ... bitmap.IP_600.70.name="logo-600" />
  <IP_4000 ... bitmap.IP_4000.70.name="logo-4000" />
```

```
</bitmaps>
```

Next, enable the idle display feature and modify the idle display "animation" for each model to point to your bitmap (again without the **.bmp** extension):

```
<indicators ind.idleDisplay.enabled="1">
  <Animations>
    <IP_300>
    ...
    </IP_300>
    <IP_330>
    ...
    <IDLE_DISPLAY ind.anim.IP_3300.38.frame.1.bitmap="logo-330"
ind.anim.IP_330.38.frame.1.duration="0"/>
    ...
    </IP_330>
    <IP_500>
    ...
    <IDLE_DISPLAY ind.anim.IP_500.38.frame.1.bitmap="logo-500"
ind.anim.IP_500.38.frame.1.duration="0"/>
    ...
    </IP_500>
    <IP_600>
    ...
    <IDLE_DISPLAY ind.anim.IP_600.38.frame.1.bitmap="logo-600"
ind.anim.IP_600.38.frame.1.duration="0"/>
    ...
    </IP_600>
    <IP_4000>
    ...
    <IDLE_DISPLAY ind.anim.IP_4000.38.frame.1.bitmap="logo-4000"
ind.anim.IP_4000.38.frame.1.duration="0"/>
    ...
    </IP_4000>
  </Animations>
  ...
</indicators>
```

BootROM/SIP Application Dependencies

Notwithstanding the hardware backward compatibility mandate, there have been times throughout the life of the SoundPoint IP / SoundStation IP phones where certain dependencies on specific bootROM and application versions have been necessitated.

This table summarizes some the major dependences that you are likely to encounter:

Model	BootROM	SIP Application
IP 301/501	2.6.1 or later	1.4.2, 1.5.4, 1.6.1 or later
IP 320/330	3.2.3 B or later	2.1.1 or later
IP 430	3.1.3 C or later	1.6.6 or later
IP 550	3.2.2 B or later	2.1 or later
IP 560	4.0.1 or later	2.2.2 or later
IP 600	2.0 or later	1.0 or later
IP 601/EM	3.1 or later	1.6 or later
IP 650/EM	3.2.2 B or later	2.0.3 B or later
IP 650/BEM	4.0.1 or later	2.2.2 or later
IP 4000	2.6 or later	1.4 or later

Migration Dependencies

In addition to the bootROM and application dependencies, there are certain restrictions with regard to upgrading or downgrading from one bootROM release to another bootROM release. These restrictions are typically caused by the addition of features that change the way bootROM provisioning is done, so the older version become incompatible.

There is always a way to move forward with bootROM releases, although it may be a two or three step procedure sometimes, but there are cases where it is impossible to move backward. Make special note of these cases before upgrading.

Note that:

- 1.x cannot be upgraded to any 2.x automatically
- 2.0 and 2.1 can not upgrade past 2.4
- Only 2.6 can upgrade to 3.0
- 3.0 cannot be downgraded

For example, a two step upgrade would be necessary from bootROM 2.1 to bootROM 2.5. A direct upgrade is not supported, but upgrading to bootROM 2.2 first, then upgrading to 2.5 will work.

Downgrade restrictions are limited to major releases. Going from 2.x to 1.x and from 3.x to 2.x are both impossible in the field.

Multiple Key Combinations

On SoundPoint IP and SoundStation IP phones, certain multiple key combinations can be used to reboot the phone and restore factory defaults.

For other methods for resetting and rebooting your SoundPoint IP or SoundStation IP phones, refer to “Quick Tip 18298: Resetting and Rebooting SoundPoint IP Phones” at <http://www.polycom.com/support/voice/>.

Rebooting the Phone

For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously until a confirmation tone is heard or for about three seconds :

- IP 301: Volume-, Volume+, Hold, and Do Not Disturb
- IP 320 and 330: Volume-, Volume+, Hold, and Hands-free
- IP 430 and 501: Volume-, Volume+, Hold, and Messages
- IP 550, 560, 600, 601, and 650: Volume-, Volume+, Mute, and Messages
- IP 4000: *, #, Volume+, and Select

Restoring Factory Defaults

For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously during the countdown process in the bootROM until the password prompt appears:

- IP 301, 501, 550, 600, 601, and 650: 4, 6, 8 and * dial pad keys
- IP 320, 330, and 430, 560: 1, 3, 5, and 7 dial pad keys
- IP 4000: 6, 8 and * dial pad keys

Enter the administrator password to initiate the reset. Resetting to factory defaults will also reset the administrator password (factory default password is 456).

Uploading Log Files

For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously until a confirmation tone is heard or for about three seconds:

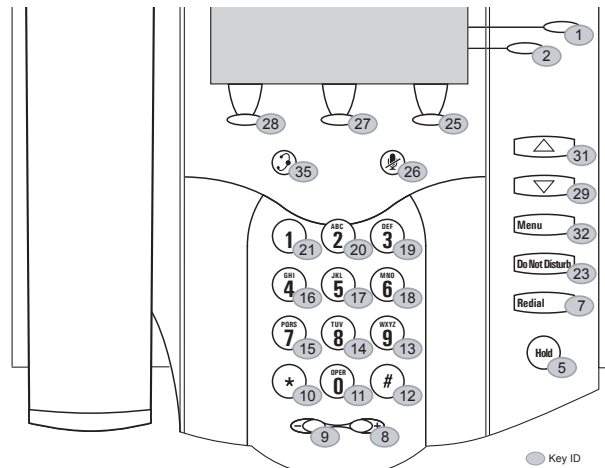
- IP 301: The two Line keys and the Up and Down arrow keys
- IP 320 and 330: Menu, Dial, and the two Line keys
- IP 430, 501, 550, 560, 600, 601, and 650: Up, Down, Left, and Right arrow keys

- IP 4000: Menu, Exit, Off-hook/Hands-free, Redial

Default Feature Key Layouts

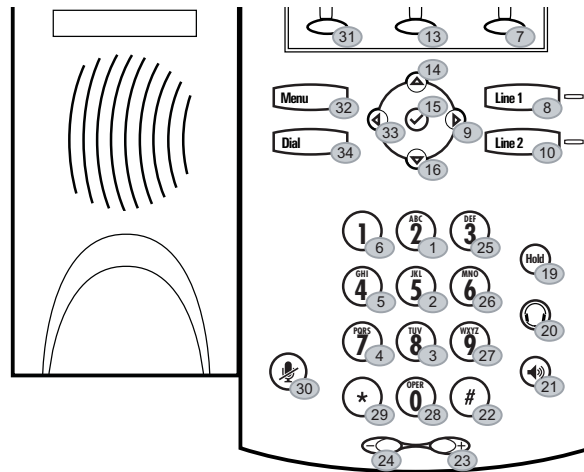
The following figures and tables show the default SIP key layouts for the SoundPoint IP 301, 320, 330, 430, 501, 550, 560, 600, 601, and 650 and SoundStation IP 4000 models.

SoundPoint IP 301



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Line1	12	DialpadPound	23	DoNotDisturb	34	n/a
2	Line2	13	Dialpad9	24	n/a	35	Headset
3	n/a	14	Dialpad8	25	SoftKey3	36	n/a
4	n/a	15	Dialpad7	26	MicMute	37	n/a
5	Hold	16	Dialpad4	27	SoftKey2	38	n/a
6	n/a	17	Dialpad5	28	SoftKey1	39	n/a
7	Redial	18	Dialpad6	29	ArrowDown	40	n/a
8	VolUp	19	Dialpad3	30	n/a	41	n/a
9	VolDown	20	Dialpad2	31	ArrowUp	42	n/a
10	DialpadStar	21	Dialpad1	32	Menu		
11	Dialpad0	22	n/a	33	n/a		

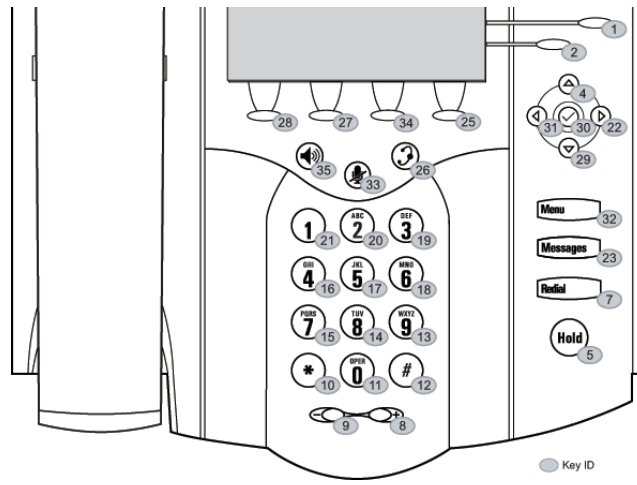
SoundPoint IP 320/330



● Key ID

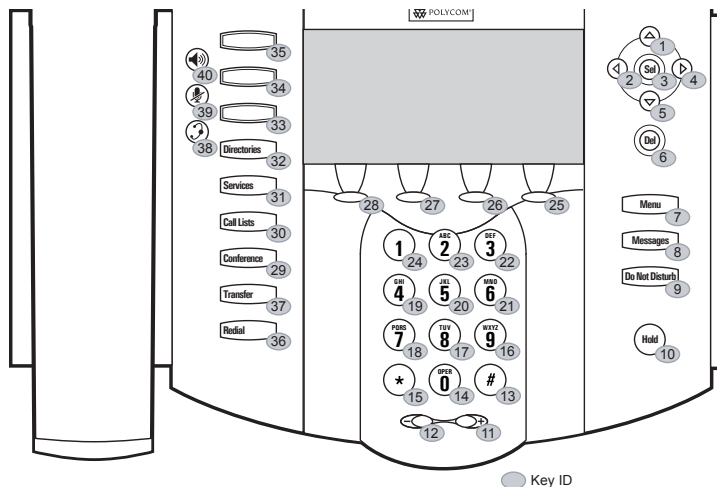
Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Dialpad2	12	n/a	23	VolUp	34	Menu
2	Dialpad5	13	SoftKey2	24	VolDown	35	n/a
3	Dialpad8	14	ArrowUp	25	Dialpad3	36	n/a
4	Dialpad7	15	Select	26	Dialpad6	37	n/a
5	Dialpad4	16	ArrowDown	27	Dialpad9	38	n/a
6	Dialpad1	17	n/a	28	Dialpad0	39	n/a
7	SoftKey3	18	n/a	29	DialpadStar	40	n/a
8	Line1	19	Hold	30	MicMute	41	n/a
9	ArrowRight	20	Headset	31	SoftKey1	42	n/a
10	Line2	21	Handsfree	32	Dial		
11	n/a	22	DialpadPound	33	ArrowLeft		

SoundPoint IP 430



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Line1	12	DialpadPound	23	Messages	34	Softkey3
2	Line2	13	Dialpad9	24	n/a	35	Handsfree
3	n/a	14	Dialpad8	25	SoftKey4	36	n/a
4	ArrowUp	15	Dialpad7	26	Headset	37	n/a
5	Hold	16	Dialpad4	27	SoftKey2	38	n/a
6	n/a	17	Dialpad5	28	SoftKey1	39	n/a
7	Redial	18	Dialpad6	29	ArrowDown	40	n/a
8	VolUp	19	Dialpad3	30	Select	41	n/a
9	VolDown	20	Dialpad2	31	ArrowLeft	42	n/a
10	DialpadStar	21	Dialpad1	32	Menu		
11	Dialpad0	22	ArrowRight	33	MicMute		

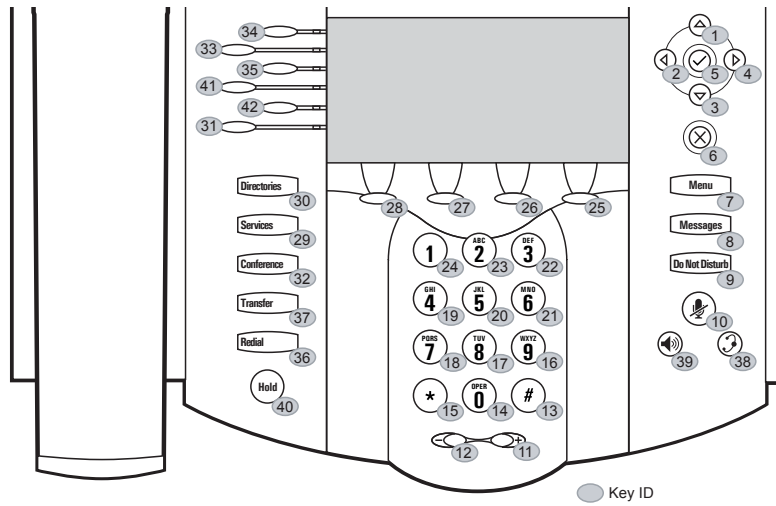
SoundPoint IP 501



● Key ID

Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	ArrowUp	12	VolDown	23	Dialpad2	34	Line2
2	ArrowLeft	13	DialpadPound	24	Dialpad1	35	Line1
3	Select	14	Dialpad0	25	SoftKey4	36	Redial
4	ArrowRight	15	DialpadStar	26	SoftKey3	37	Transfer
5	ArrowDown	16	Dialpad9	27	SoftKey2	38	Headset
6	Delete	17	Dialpad8	28	SoftKey1	39	MicMute
7	Menu	18	Dialpad7	29	Conference	40	Handsfree
8	Messages	19	Dialpad4	30	CallHistory	41	n/a
9	DoNotDisturb	20	Dialpad5	31	Services	42	n/a
10	Hold	21	Dialpad6	32	Directories		
11	VolUp	22	Dialpad3	33	Line3		

SoundPoint IP 550/560/600/601/650

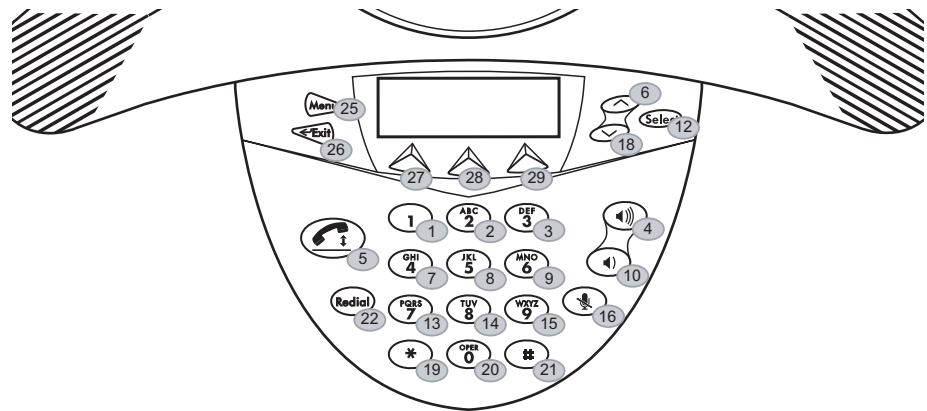


Note

The SoundPoint IP 550 and 560 has have only the top four lines keys. Key IDs 31 and 42 are not used on SoundPoint IP 550 and 560 phones.

Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	ArrowUp	12	VolDown	23	Dialpad2	34	Line1
2	ArrowLeft	13	DialpadPound	24	Dialpad1	35	Line3
3	ArrowDown	14	Dialpad0	25	SoftKey4	36	Redial
4	ArrowRight	15	DialpadStar	26	SoftKey3	37	Transfer
5	Select	16	Dialpad9	27	SoftKey2	38	Headset
6	Delete	17	Dialpad8	28	SoftKey1	39	Handsfree
7	Menu	18	Dialpad7	29	Services	40	Hold
8	Messages	19	Dialpad4	30	Directories	41	Line4
9	DoNotDisturb	20	Dialpad5	31	Line6	42	Line5
10	MicMute	21	Dialpad6	32	Conference		
11	VolUp	22	Dialpad3	33	Line2		

SoundStation IP 4000



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Dialpad1	12	Select	23	n/a	34	n/a
2	Dialpad2	13	Dialpad7	24	n/a	35	n/a
3	Dialpad3	14	Dialpad8	25	Menu	36	n/a
4	VoIUp	15	Dialpad9	26	Exit	37	n/a
5	Handsfree	16	MicMute	27	SoftKey1	38	n/a
6	ArrowUp	17	n/a	28	SoftKey2	39	n/a
7	Dialpad4	18	ArrowDown	29	SoftKey3	40	n/a
8	Dialpad5	19	DialpadStar	30	n/a	41	n/a
9	Dialpad6	20	Dialpad0	31	n/a	42	n/a
10	VoIDown	21	DialpadPound	32	n/a		
11	n/a	22	Redial	33	n/a		

Assigning a VLAN ID Using DHCP

To assign a VLAN ID to a phone using DHCP:

>> In the DHCP menu of the Main setup menu, set *VLAN Discovery* to "Fixed" or "Custom".

When set to "Fixed", the phone will examine DHCP options 128,144, 157 and 191 (in that order) for a valid DVD string.

When set to "Custom", the value set in "VLAN ID Option" will be examined for a valid DVD string.

DVD string in the DHCP option must meet the following conditions to be valid:

- Must start with ?VLAN-A=? (case-sensitive)
- Must contain at least one valid ID
- VLAN IDs range from 0 to 4095
- Each VLAN ID must be separated by a ?+? character
- The string must be terminated by a ;?
- All characters after the ;? will be ignored
- There must be no white space before the ;?
- VLAN IDs may be decimal, hex, or octal

For example:

The following DVD strings will result in the phone using VLAN 10:

VLAN-A=10;

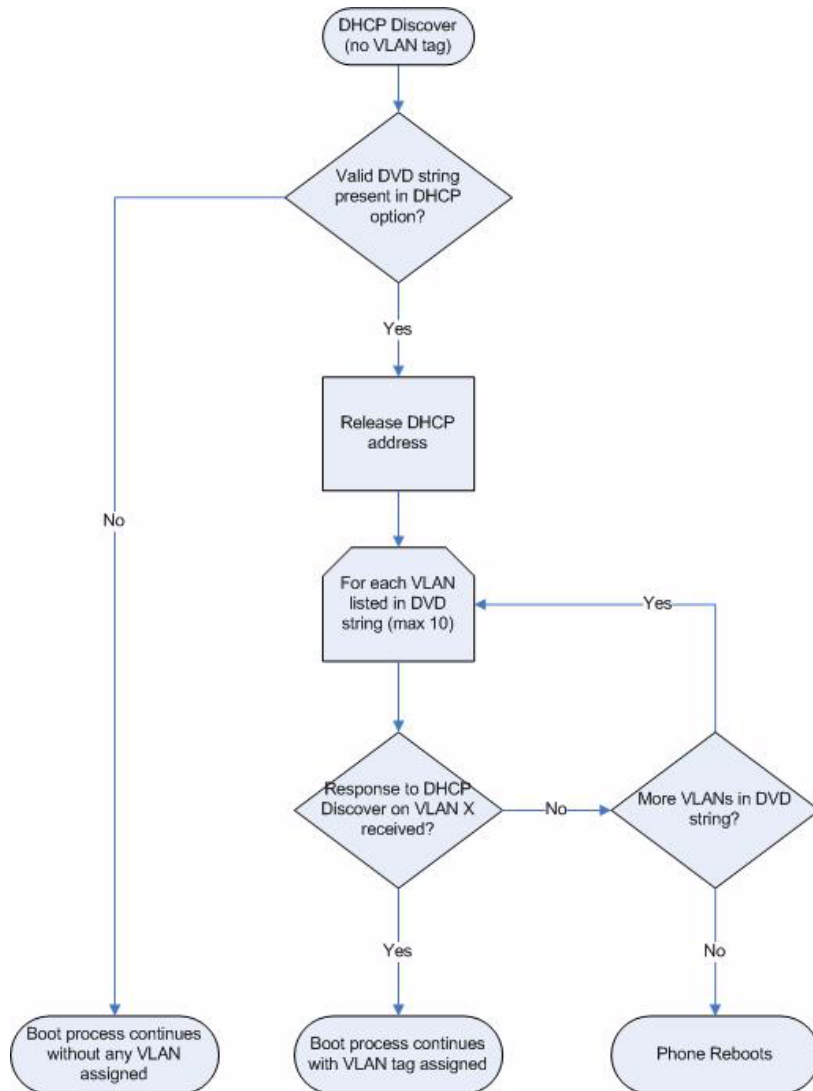
VLAN-A=0x0a;

VLAN-A=012;

Note

If a VLAN tag is assigned by CDP, DHCP VLAN tags will be ignored.

The following figure shows the phone's processing to determine if the VLAN ID is valid:



Parsing Vendor ID Information

After the phone boot, it sends a DHCP Discover packet to the DHCP server. This is found in the Bootstrap Protocol/option "Vendor Class Identifier" section of the packet and includes the phone's part number and the bootROM version. The format of this option's data is not specified in RFC 2132, but is left to each vendor to define its own format. To be useful, every vendor's format must be distinguishable from every other vendor's format. To make our format uniquely identifiable, the format follows RFC 3925, which uses the

IANA Private Enterprise number to determine which vendor's format should be used to decode the remaining data. The private enterprise number assigned to Polycom is 13885 (0x0000363D).

This vendor ID information is not a character string, but an array of binary data. The steps for parsing are as follows:

1. Check for the Polycom signature at the start of the option:
4 octet: 00 00 36 3d
2. Get the length of the entire list of sub-options:
1 octet
3. Read the field code and length of the first sub-option, 1+1 octets
4. If this is a field you want to parse, save the data.
5. Skip to the start of the next sub-option.
6. Repeat steps 3 to 5 until you have all the data or you encounter the End-of-Suboptions code (0xFF).

For example, the following is a sample decode of a packet from an IP601:

```

3c 74
  - Option 60, length of Option data (part of the DHCP spec.)
00 00 36 3d
  - Polycom signature (always 4 octets)
6f
  - Length of Polycom data
01 07 50 6f 6c 79 63 6f 6d
  - sub-option 1 (company), length, "Polycom"
02 15 53 6f 75 6e 64 50 6f 69 6e 74 49 50 2d 53 50 49 50 5f 36 30 31
  - sub-option 2 (part), length, "SoundPointIP-SPIP_601"
03 10 32 33 34 35 2d 31 31 36 30 35 2d 30 30 31 2c 32
  - sub-option 3 (part number), length, "2345-11605-001,2"
04 1c 53 49 50 2f 54 69 70 2e 58 58 58 58 2f 30 38 2d 4a 75 6e 2d 30 37
20 31 30 3a 34 34
  - sub-option 4 (Application version), length, "SIP/Tip.XXXX/08-Jun-07
10:44"
05 1d 42 52 2f 33 2e 31 2e 30 2e 58 58 58 58 2f 32 38 2d 41 70 72 2d 30
35 20 31 33 3a 33 30
  - sub-option 5 (BootROM version), length, "BR/3.1.0.XXXX/28-Apr-05
13:30"
ff
  - end of sub-options

```

For the BootROM, sub-option 4 and sub-option 5 will contain the same string. The string is formatted as follows:

```
<apptype>/<buildid>/<date+time>
```

where:

```
<apptype> can be 'BR' (BootROM) or 'SIP' (SIP Application)
```

Third Party Software

This appendix provides the copyright statements for third party software products that are part of the Session Initiation Protocol (SIP) application.

Ares

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zlib

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Jean-loup Gailly Mark Adler
jloup@zip.org madler@alumni.caltech.edu

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curl

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Addendum to SIP 3.0 Administrator's Guide

This addendum addresses changes to the SoundPoint IP / SoundStation IP SIP 3.0 Administrator's Guide specific to the release of the SoundStation IP 7000 conference phone.

The SoundStation IP 7000 conference phone is a next-generation phone with a modern industrial design and the latest advancements in Polycom voice technology. Some of the features include:

- Extraordinary audio performance
- Strong IP telephony feature set
- Built-in voice bridge for multi-party conferences
- Ability to connect multiple consoles and microphones together for convenience, performance, and flexibility
- A port built into the console to allow calls from a mobile phone
- Daisy-chaining
- Provisioning over CLink

The new or changed features include:

- [Distribution Zip File](#)
- [Managing Conferences](#)
- [Configurable Feature Keys](#)
- [Multilingual User Interface](#)
- [Single Registration](#)
- [Audio Codecs](#)
- [Treble/Bass Controls](#)
- [Voice Quality Monitoring](#)

- [Daisy-Chaining Phones](#)
- [Provisioning Phones Over CLink](#)

Configuration file changes are described in [Configuration File Changes](#) on page 1-5.

The default SIP key layout for the SoundStation IP 7000 conference phone is shown in [Multiple Key Combinations and Default Key Layout](#) on page 1-12.

For more information, refer to the Release Notes for the SIP Application, Version 3.0.2 .

Note

For more information on the SoundStation IP 7000 conference phone, refer to the *User Guide* at <http://www.polycom.com/support/voip/> .

New or Changed Features

Distribution Zip File

As well as the **sip.ld** file in the archive, there is a separate file for the SoundStation IP 7000 conference phone called **3111-40000-001.sip.ld** .

Managing Conferences

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Turn this feature on or off. <ul style="list-style-type: none">• For more information, refer to Feature <feature/> on page 1-12.
----------------------------------	---------------------------------------	----------------------------------------------------------------------------------------------------------------------------------------------------------------------

Configurable Feature Keys

No feature keys on the SoundStation IP 7000 can be remapped.



Multilingual User Interface

Support for major western European languages is included and additional languages can be easily added. Support for Asian languages (Chinese, Japanese, and Korean) is also included and will display on the SoundStation IP 7000's higher resolution display. A WGL4 character set is displayed the SoundStation IP 7000. For more information, refer to <http://www.microsoft.com/OpenType/otspec/WGL4E.HTM>. Downloadable fonts are not supported on the SoundStation IP 7000.

Single Registration

The SoundStation IP 7000 phone supports a single registration.

When the phone is unable to register with the call control server, the icon

 is shown (outline). Once the phone is able to register, the icon  is shown (solid).

Audio Codecs

The following table summarizes the SoundStation IP 7000 phone's new audio codec support:

Algorithm	MIME Type	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
G.722.1C	G7221/ 32000	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz
Lin16	L16/16000 L16/32000 L16/48000	25.6 Kbps 51.2 Kbps 76.8 Kbps	16 Ksps 32 Ksps 48 Ksps	10ms	7 KHz 14 KHz 22 KHz
Siren14	SIREN14/ 16000	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz
Siren22	SIREN22/ 48000	32 Kbps 48 Kbps 64 Kbps	32 Ksps	20ms - 80ms	14 KHz

Note

The network bandwidth necessary to send the encoded voice is typically 5-10% (confirm number) higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps consumes 5xkbps of network bandwidth.

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: sip.cfg	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. <ul style="list-style-type: none"> For more information, refer to Codec Preferences <codecPref/> on page 1-8 and Codec Profiles <audioProfile/> on page 1-9.
----------------------------------	---------------------------------------	---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

Treble/Bass Controls

The treble and bass controls equalize the tone of the high and low frequency sound from the speakers.

The SoundStation IP 7000 phone's treble and bass controls can be modified by the user (through **Menu > Settings > Basic > Audio > Treble EQ** or **Bass EQ**).

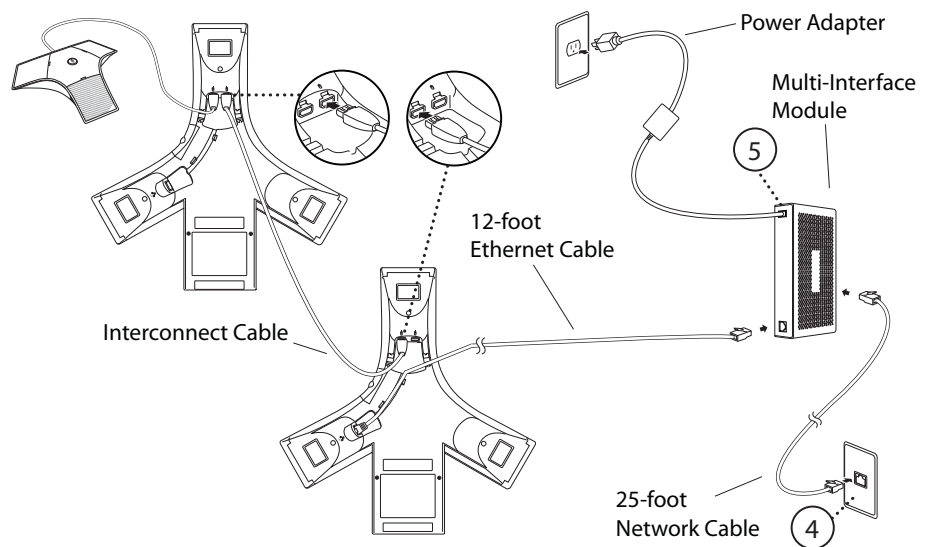
There are no related configuration changes.

Voice Quality Monitoring

Voice Quality Monitoring is not supported on the SoundStation IP 7000 conference phone at this time.

Daisy-Chaining Phones

You can join two SoundStation IP 7000 conference phones together through the use of a CLink cable and the Multi-Interface Module. The graphic display of each phone shows the same user interface and phone numbers. The SoundStation IP 7000 phone that has the Ethernet connection is referred to as the primary. The SoundStation IP 7000 phone that does not have the Ethernet connection is referred to as the secondary. The primary/secondary relationship of the phones is determined by their MAC address, registration status, and the configuration files.



Instructions for daisy-chaining SoundStation IP 7000 conference phones are available in the *SoundStation IP 7000 User Guide*.

Provisioning Phones Over CLink

Normally the SoundStation IP 7000 conference phone is provisioned over the Ethernet by the boot server. However, when two SoundStation IP 7000 phones are daisy-chained together, the one that is not directly connected to the Ethernet can still be provisioned (known as the secondary). The provisioning over CLink feature is automatically enabled when a SoundStation IP 7000 phone is not connected to the Ethernet. Both SoundStation IP 7000 phones must be running the same version of the SIP application.

The steps for provisioning the secondary SoundStation IP 7000 phone are the same as for the primary SoundStation IP 7000 phone. You can reboot the primary without rebooting the secondary. However, the primary and secondary should be rebooted together for the primary/secondary relationship to be recognized. If you power up both SoundStation IP 7000 phones, the primary will power up first.

Currently, provisioning over CLink is supported for the following configurations of SoundStation IP 7000 conference phones:

- Two SoundStation IP 7000 conference phone daisy-chained together
- Two SoundStation IP 7000 conference phone daisy-chained together with one external microphone, specifically designed for the SoundStation IP 7000 conference phone

Refer to [Daisy-Chaining Phones](#) on page 1-4 for an illustration of two SoundStation IP 7000 conference phone daisy-chained together.

The provisioning boot server (or proxy) for the secondary is determined by the following criteria:

- If the secondary is configured for DHCP, use the primary's boot server if the primary is configured for DHCP.
- If the secondary is not configured for DHCP, use the secondary's static boot server if it exists.
- If the secondary's static boot server does not exist, use the primary's boot server (ignoring the source).

Configuration File Changes

The following `sip.cfg` configuration file changes were made to support the SoundStation IP 7000 conference phone:

- `SDP <SDP/>`
- `User Preferences <up/>`
- `Sampled Audio for Sound Effects <saf/>`
- `Voice Coding Algorithms <codecs/>`

- Gains <gain/>
- Receive Equalization <rxEq/>
- Transmit Equalization <txEq/>
- Call Handling Configuration <call/>
- Bitmaps <bitmap/>
- Feature <feature/>

SDP <SDP/>

This new configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
volpProt.SDP.useLegacyPayloadNegotiation	0, 1	Null	If set to 1, the phone transmits and receives RTP using the payload type identified by the first codec listed in the SDP of the codec negotiation answer. If set to 0 or Null, RFC 3264 is followed for transmit and receive RTP payload type values.

User Preferences <up/>

These configuration attributes are defined as follows:

Attribute	Permitted Values	Default	Interpretation
up.toneControl.bass	-4 to 4, Null	0	Bass equalization control. Each step is an increment of 1 dB at 225 kHz and 2 dB < 225 Hz.
up.toneControl.treble	-4 to 4, Null	0	Treble equalization control. Each step is an increment of 1 dB at 3.7 kHz and 2 dB > 10 kHz.

Attribute	Permitted Values	Default	Interpretation
up.audioSetup.auxInput	0 - Other Input, 1 - Polycom Wireless Mic, 2 - off	Null	Auxiliary audio input. If set to Null, default value is 2.
up.audioSetup.auxOutput	0 - Other Input, 1 - Polycom Wireless Mic, 2 - off	Null	Auxiliary audio output. If set to Null, default value is 2.

Sampled Audio for Sound Effects <sa/>

The following new sampled audio WAVE file (.wav) formats are supported:

- L16/32000 (16-bit, 32 kHz sampling rate, mono)
- L16/48000 (16-bit, 48 kHz sampling rate, mono)

Voice Coding Algorithms <codecs/>

These new codecs include:

- [Codec Preferences <codecPref/>](#)
- [Codec Profiles <audioProfile/>](#)

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_7000.G711Mu	Null, 1-16	6	Specifies the codec preferences for the SoundStation IP 7000 platform. Interpretation as above.
voice.codecPref.IP_7000.G711A		7	
voice.codecPref.IP_7000.G722		4	
voice.codecPref.IP_7000.G7221.16kbps		Null	
voice.codecPref.IP_7000.G7221.24kbps		Null	
voice.codecPref.IP_7000.G7221.32kbps		5	
voice.codecPref.IP_7000.G7221C.24kbp s		Null	
voice.codecPref.IP_7000.G7221C.32kbp s		Null	
voice.codecPref.IP_7000.G7221C.48kbp s		2	
voice.codecPref.IP_7000.G729AB		8	
voice.codecPref.IP_7000.Lin16.16ksps		Null	
voice.codecPref.IP_7000.Lin16.32ksps		Null	
voice.codecPref.IP_7000.Lin16.48ksps		Null	
voice.codecPref.IP_7000.Siren22.32kbp s		Null	
voice.codecPref.IP_7000.Siren22.48kbp s		Null	
voice.codecPref.IP_7000.Siren22.64kbp s		1	
voice.codecPref.IP_7000.Siren14.24kbp s		Null	
voice.codecPref.IP_7000.Siren14.32kbp s		Null	
voice.codecPref.IP_7000.Siren14.48kbp s		3	

Note

Codecs with a default of Null are available for test purposes only and are not expected to be used in your deployment.

Codec Profiles <audioProfile/>

The profile attributes can be adjusted for each of the new supported codecs, namely, $x = G7221C, Lin16, Siren14,$ and $Siren22.$

Attribute	Permitted Values	Interpretation
voice.audioProfile.x.payloadSize	10, 20, 30, ...80	Preferred Tx payload size in milliseconds to be provided in SDP offers and used in the absence ofptime negotiations. This is also the range of supported Rx payload sizes. The payload size for G7221, G7221C, Siren14, and Siren22 are further subdivided.
voice.audioProfile.x.jitterBufferMin	20, 40, 50, 60, ... (multiple of 10)	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter. The IP4000 values are the same as the IP30x values.
voice.audioProfile.x.jitterBufferShrink	10, 20, 30, ... (multiple of 10)	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (1000 ms) to minimize the delay on known good networks. Use larger values to minimize packet loss on networks with large jitter (3000 ms).
voice.audioProfile.x.jitterBufferMax	> jitterBufferMin, multiple of 10, <=300 for IP 320, 330, 430, 501,550, 600, 601, and 650 <= 200 for IP 301	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.

Gains <gain/>

The default gain settings have been carefully adjusted to comply with the TIA-810-A digital telephony standard.



Polycom recommends that you do not change these values.

Attribute	Default
voice.gain.rx.analog.chassis.IP_7000	0
voice.gain.rx.analog.ringer.IP_7000	0
voice.gain.rx.digital.chassis.IP_7000	0
voice.gain.rx.digital.ringer.IP_7000	-21
voice.gain.tx.analog.chassis.IP_7000	0
voice.gain.tx.digital.chassis.IP_7000	6

Receive Equalization <rxEq/>

These settings control the performance of the receive equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.rxEq.hf.IP_7000.preFilter.enable	0
voice.rxEq.hf.IP_7000.postFilter.enable	0

Transmit Equalization <txEq/>

These settings control the performance of the hands-free transmit equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.txEq.hf.IP_7000.preFilter.enable	0
voice.txEq.hf.IP_7000.postFilter.enable	0

Call Handling Configuration <call/>

These new configuration attributes are defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.singleKeyPressConference	0,1	0	<p>If set to 1, the conference will be setup after a user presses the Conference soft key or Conference key the first time. Also, all sound effects (dial tone, DTMF tone while dialing and ringing back) are heard by all existing participants in the conference.</p> <p>If set to 0 or Null, sound effects are only heard by conference initiator (old behavior).</p> <p>Only supported for SoundPoint IP 550, 560, and 650 and SoundStation IP 7000. For all others, set to 0.</p>
call.localConferenceCallHold	0, 1	0	<p>If set to 0, a hold will happen for all legs when conference is put on hold. (old behavior).</p> <p>If set to 1, only the host is out of the conference, all other parties in conference continue to talk. (new behavior).</p> <p>If set to Null, the default value is 0.</p> <p>Only supported for SoundPoint IP 550, 560, and 650 and SoundStation IP 7000. For all others, set to 0.</p>

Attribute	Permitted Values	Default	Interpretation
call.cellPhoneAutoBridging	Disabled, Enabled	Disabled	<p>If set to Enabled, the cell phone audio is automatically bridged to active SIP call (conference call). If one SIP call is put on hold and another SIP call is activated the cell phone audio is automatically bridged to the newly active SIP call.</p> <p>If set to Disabled, the cell phone audio is available only if there is no active SIP calls. If all SIP calls are placed on hold, then the cell phone audio is available.</p>

Bitmaps <bitmap/>

The IP_7000 parameters affect the SoundStation IP 7000 conference phone.

Indicators <ind/>

The IP_7000 parameters affect the SoundStation IP 7000 conference phone.

Animations <anim/>

The IP_7000 parameters affect the SoundStation IP 7000 conference phone.

Graphic Icons <gi/>

The IP_7000 parameters affect the SoundStation IP 7000 conference phone.

Feature <feature/>

The “call list” feature cannot be disabled on the SoundStation IP 7000 phone.

The manage conference feature is always enabled on the SoundStation IP 7000 phone.

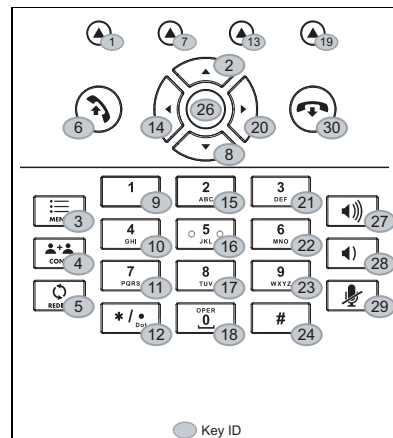
Multiple Key Combinations and Default Key Layout

The multiple key combinations on the SoundStation IP 7000 conference phone are as follows:

- Rebooting the Phone—Press and hold *, #, Volume-, and Volume+ simultaneously until a confirmation tone is heard or for about three seconds.

- Restoring Factory Defaults – Press and hold 1, 3, 5, and 7 dial pad keys simultaneously during the countdown process in the bootROM until the password prompt appears. Enter the administrator password to initiate the reset. Resetting to factory defaults will also reset the administrator password (factory default password is 456).
- Uploading Log Files – Press and hold Up, Down, Left, and Right arrow keys simultaneously until a confirmation tone is heard or for about three seconds.

The SoundStation IP 7000 conference phone default key layout is as follows:



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	SoftKey1	12	DialpadStar	23	Dialpad9	34	n/a
2	ArrowUp	13	SoftKey3	24	DialpadPound	35	n/a
3	Menu	14	ArrowLeft	25	n/a	36	n/a
4	Conference	15	Dialpad2	26	Select	37	n/a
5	Redial	16	Dialpad5	27	VoUp	38	n/a
6	Handsfree	17	Dialpad8	28	VoDown	39	n/a
7	SoftKey2	18	Dialpad0	29	MicMute	40	n/a
8	ArrowDown	19	SoftKey4	30	Release	41	n/a
9	Dialpad1	20	ArrowRight	31	n/a	42	n/a
10	Dialpad4	21	Dialpad3	32	n/a		
11	Dialpad7	22	Dialpad6	33	n/a		

Addendum to SIP 3.0 Administrator's Guide

This addendum addresses changes to the SoundPoint IP / SoundStation IP SIP 3.0 Administrator's Guide specific to the release of the SoundStation IP 6000 conference phone.

The SoundPoint IP 6000 conference phone is a next-generation phone with a modern industrial design and the latest advancements in Polycom voice technology. Some of the features include:

- Excellent wideband audio performance
- A console microphone with a range of up to ten feet
- Expansion microphones to accommodate large room sizes

The new or changed features include:

- [Distribution Zip File](#)
- [Configurable Feature Keys](#)
- [Multilingual User Interface](#)
- [Single Registration](#)
- [Audio Codecs](#)
- [Voice Quality Monitoring](#)

Configuration file changes are described in [Configuration File Changes](#) on page 1-3.

The default SIP key layout for the SoundPoint IP 6000 conference phone is shown in [Multiple Key Combinations and Default Key Layout](#) on page 1-6.

For more information, refer to the Release Notes for the SIP Application, Version 3.0.2 .

Note

For more information on the SoundPoint IP 6000 conference phone, refer to the *User Guide* at <http://www.polycom.com/support/voip/> .

New or Changed Features

Distribution Zip File

As well as the **sip.ld** file in the archive, there is a separate file for the SoundPoint IP 6000 conference phone called **3111-15600-001.sip.ld**.

Configurable Feature Keys

No feature keys on the SoundStation IP 6000 can be remapped.

Multilingual User Interface

Support for major western European languages is included and additional languages can be easily added. Support for Asian languages (Chinese, Japanese, and Korean) is also included and will display on the SoundStation IP 6000's higher resolution display. Downloadable fonts are not supported on the SoundStation IP 6000.

Single Registration

The SoundStation IP 6000 phone supports a single registration.

When the phone is unable to register with the call control server, the icon

 is shown (outline). Once the phone is registered, the icon  is shown (solid).

Audio Codecs

The following table summarizes the SoundStation IP 6000 phone's audio codec support:

Algorithm	MIME Type	Ref.	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
G.722.1C	G7221/ 32000	G7221C	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz
Siren14	SIREN14/ 16000	SIREN14	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz

Note

The network bandwidth necessary to send the encoded voice is typically 5-10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps consumes 5xkbps of network bandwidth (one-way audio). Two-way audio would take over 100kbps.

Configuration changes can be performed centrally at the boot server or locally:

Central (boot server)	Configuration file: sip.cfg	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. <ul style="list-style-type: none"> For more information, refer to Codec Preferences <codecPref/> on page 1-4.
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Voice Quality Monitoring

Voice Quality Monitoring is not supported on the SoundStation IP 6000 conference phone at this time.

Configuration File Changes

The following **sip.cfg** configuration file changes were made to support the SoundStation IP 6000 conference phone:

- [Sampled Audio for Sound Effects <saf/>](#)
- [Voice Coding Algorithms <codecs/>](#)
- [Gains <gain/>](#)
- [Receive Equalization <rxEq/>](#)
- [Transmit Equalization <txEq/>](#)
- [Feature <feature/>](#)

Sampled Audio for Sound Effects <saf/>

The following new sampled audio WAVE file (.wav) formats are supported:

- L16/32000 (16-bit, 32 kHz sampling rate, mono)
- L16/48000 (16-bit, 48 kHz sampling rate, mono)

Voice Coding Algorithms <codecs/>

The codecs include:

- [Codec Preferences <codecPref/>](#)

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_6000.G711Mu	Null, 1-13	5	Specifies the codec preferences for the SoundStation IP 6000 platform. Interpretation as above.
voice.codecPref.IP_6000.G711A		6	
voice.codecPref.IP_6000.G722		3	
voice.codecPref.IP_6000.G7221.16kbps		Null	
voice.codecPref.IP_6000.G7221.24kbps		Null	
voice.codecPref.IP_6000.G7221.32kbps		4	
voice.codecPref.IP_6000.G729AB		7	
voice.codecPref.IP_6000.G7221C.24kbp s		Null	
voice.codecPref.IP_6000.G7221C.32kbp s		Null	
voice.codecPref.IP_6000.G7221C.48kbp s		1	
voice.codecPref.IP_6000.Siren14.24kbp s		Null	
voice.codecPref.IP_6000.Siren14.32kbp s		Null	
voice.codecPref.IP_6000.Siren14.48kbp s		2	

Note

Codecs with a default of Null are available for test purposes only and are not expected to be used in your deployment.

Gains <gain/>

The default gain settings have been carefully adjusted to comply with the TIA-810-A digital telephony standard.



Polycom recommends that you do not change these values.

Attribute	Default
voice.gain.rx.analog.chassis.IP_6000	0
voice.gain.rx.analog.ringer.IP_6000	0
voice.gain.rx.digital.chassis.IP_6000	6
voice.gain.rx.digital.ringer.IP_6000	-21
voice.gain.tx.analog.chassis.IP_6000	0
voice.gain.tx.digital.chassis.IP_6000	6

Receive Equalization <rxEq/>

These settings control the performance of the receive equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.rxEq.hf.IP_6000.preFilter.enable	0
voice.rxEq.hf.IP_6000.postFilter.enable	0

Transmit Equalization <txEq/>

These settings control the performance of the hands-free transmit equalization feature.



Polycom recommends that you do not change these values.

Attribute	Default
voice.txEq.hf.IP_6000.preFilter.enable	0
voice.txEq.hf.IP_6000.postFilter.enable	0

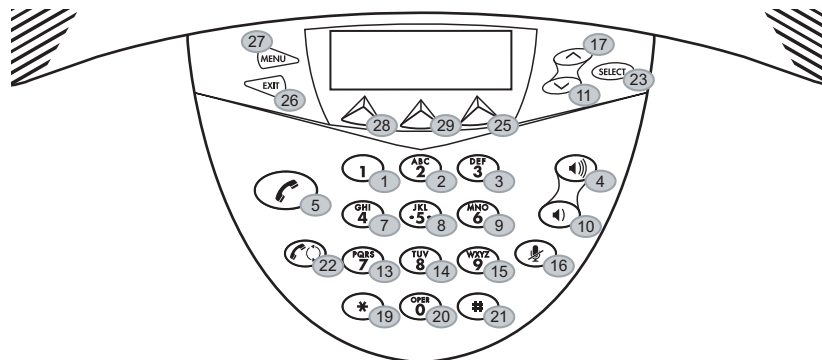
Feature <feature/>

The “call list” feature cannot be disabled on the SoundStation IP 6000 phone.

Multiple Key Combinations and Default Key Layout

The multiple key combinations on the SoundStation IP 6000 conference phone are the same as the SoundStation IP 4000 conference phone.

The SoundStation IP 6000 conference phone default key layout is as follows:



● Key ID

Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Dialpad1	12	n/a	23	Select	34	n/a
2	Dialpad2	13	Dialpad7	24	n/a	35	n/a
3	Dialpad3	14	Dialpad8	25	SoftKey3	36	n/a
4	VolUp	15	Dialpad9	26	Exit	37	n/a

Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
5	Handsfree	16	MicMute	27	Menu	38	n/a
6	n/a	17	ArrowUp	28	SoftKey1	39	n/a
7	Dialpad4	18	n/a	29	SoftKey2	40	n/a
8	Dialpad5	19	DialpadStar	30	n/a	41	n/a
9	Dialpad6	20	Dialpad0	31	n/a	42	n/a
10	VolDown	21	DialpadPound	32	n/a		
11	ArrowDown	22	Redial	33	n/a		

